MAGIX

SAMPLITUDE PRO X7

English language manual

Copyright

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Welcome

Thank you for choosing Samplitude!

You now own one of the most successful, complete solutions for professional audio editing. This PC-based Digital Audio Workstation (DAW) features extensive application options for recording, editing, mixing, media authoring, and mastering. For over two decades this program has offered the perfect combination of unique functionality & sound neutrality, outstanding cutting & editing options, perfect CD/DVD mastering, and flexible customization of individual workflows.

Just as in the past, the current version was developed in close collaboration with musicians, sound engineers, producers, and users who consider Samplitude an absolute must for their professional productions. It offers numerous innovative functions and incorporates the advanced development of tried and tested performance features.

The amazing Audio Engine with complete bit transparency and accurate phase stability has been specially designed and optimized to meet the highest professional standards. Work with a fully customizable interface and experience a DAW tailored to your needs.

Samplitude Pro X6 and Samplitude Pro X6 Suite are now available as either a 64-bit version or 32-bit version.

We wish you lots of success with Samplitude!

Your Samplitude Team

What's New in Samplitude Pro X7?

The documentation for the new functions of Samplitude can be found in a separate PDF document **Samplitude Pro X7 New Features.pdf** in the program folder of Samplitude!

- **Improved crossfade editor**: In the new Crossfade Editor, crossfades between objects can be edited in detail, even fully keyboard controlled.
- **Redesigned Track Editor**: The Track Editor has been reorganized for better clarity, the important controls for the AUX send, track effects and automation are now together in one section.
- **Unified export dialog**: The functions in the File menu > Export... for file export to the various audio formats and the Trackbouncing dialog have been combined into a new, unified Export dialog.
- **Revised dockable mixer**: The Mixer can now be docked anywhere in the program window. The controls adapt dynamically to the available space.
- **ExternalFX plug-in**: Outboard effects can be conveniently integrated via a special ExternalFX plug-in.
- **MIDI plug-ins**: It is now possible to use MIDI plug-ins such as sequencers, arpeggiators or chord utilities together with virtual instruments on one track.
- **Dockable Plug-in Browser**: The Plug-in Browser 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.
- **Selectable input of the Visualization:** The input signal of the Visualization can now be selected directly from a menu in the visualization window.

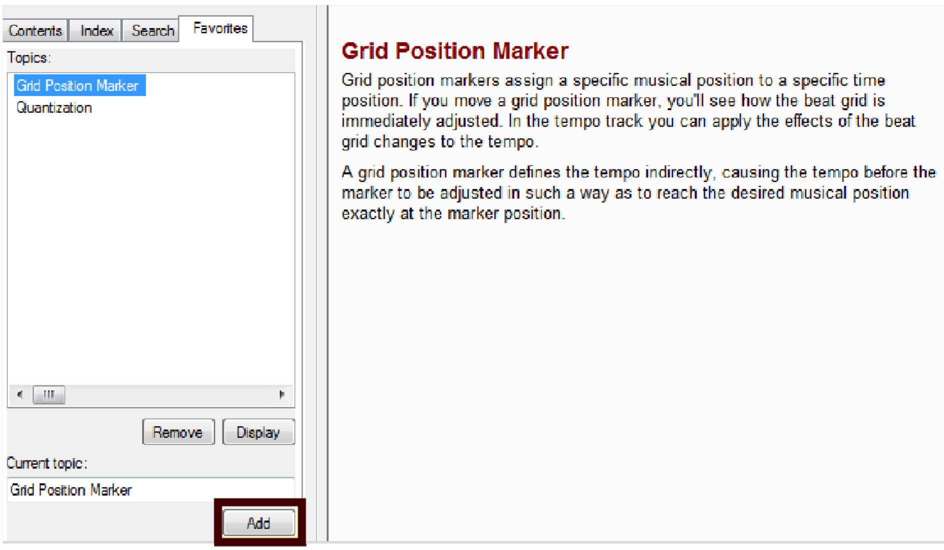
Documentation and Help

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

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

Help features

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. There you can jump to specific topics of the help via the table of contents, in the tab **Index** you can search for keywords, in the tab **Search** there is a full text search. Help topics that you need more often can be saved in the tab **Favorites**.



In addition, a search box (view page 101) in the top 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**.

Select Menu **Help** > **Context Help...** (keyboard shortcut: **Shift + F1**) and then click on an element of the program interface to get help on this control.

Support and user forum

Registered users can receive technical support by telephone or email. For more information, visit our online support area at https://www.magix.com/us/support/technical-support/pro-audio-support/.

Here's where you can find the latest downloads for your software product. Also visit the Pro Audio User Forum, which can be accessed via the website https://support2.magix.com/boards/samplitude.

As a registered user you will be supported by the professional Samplitude forum community and can get involved in all the conversations.

PDF Documentation

All information from the help can also be found in a PDF document in the program folder of <program name>. It can be opened via menu **Help** > **Manual**. This folder also contains a clear, print-ready PDF document with the most important program keyboard shortcuts.

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

Contact

Product Activation

If you have questions about product activation, get in touch with the Samplitude Service Team:

Phone: +49 (0) 5741 3455 25 (Mon - Fri from 10AM to 4PM CET)

Fax: + 49 (0)5741 3107 68

Email: infoservice@magix.net

Support

Registered customers receive technical support:
https://www.magix.com/us/support/technical-support/pro-audio-support/

System Requirements

Samplitude On Windows x64 (64-Bit)

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

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

Recommended program versions:

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

Samplitude On Windows x86 (32-Bit)

If you have installed the 32-bit version of Windows on your system you can only use the 32-bit version of Samplitude.

The 64-Bit Version Should I Switch?

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

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

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

General System Requirements

Supported operating systems

- Microsoft Windows 10 | 11 (32-Bit und 64-Bit)

Minimum system requirements

- **Processor**: 2 GHz and higher
- **RAM**: 2 GB RAM (32-bit), 4 GB RAM (64-bit)
- **Graphics card**: Onboard, min. resolution 1280 x 1024 pixels
- **Hard drive space:** 4 GB for program installation, 20 GB for Samplitude Pro X, 100 GB for Samplitude Pro X Suite
- **Internet connection**: Required for registering and validating the program, as well as for some program functions. Program requires one-time registration.
- **Audio playback**: Sound card (ASIO-enabled sound card recommended)
- **DVD**
- **Optional**: CD/DVD ±R(W) recorder, MIDI interface

Hard disk

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

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". You can view the contents of the installation CD in the installation screen, as well as visit our website or install additional programs such as CodeMeter Runtime.
2. If you have purchased a download version, run the downloaded installer.
3. Next choose the language in which you want to install the program.

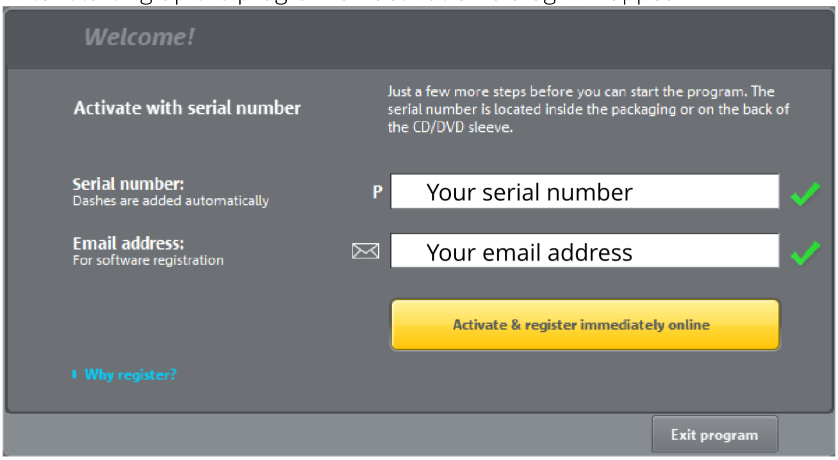
> The program documentation is only available in German and English.

4. To begin the installation of Samplitude click on "Install program" > "Samplitude".
5. Follow the instructions and click "Next" each time. If you choose the "User defined" installation style, you can also specify the target folder for the program folder and select additional components for installation. In the selection screen, you will see the total required memory for the installation.
6. After all files have been copied to the hard drive, confirm the end of the installation by clicking on "Finish".

After the initial installation, you can start the program via the Windows "Start" menu or the desktop link. After installation, you can add or remove components by starting this installation program again and selecting "Custom" to select or deselect the respective components.

Activating Samplitude

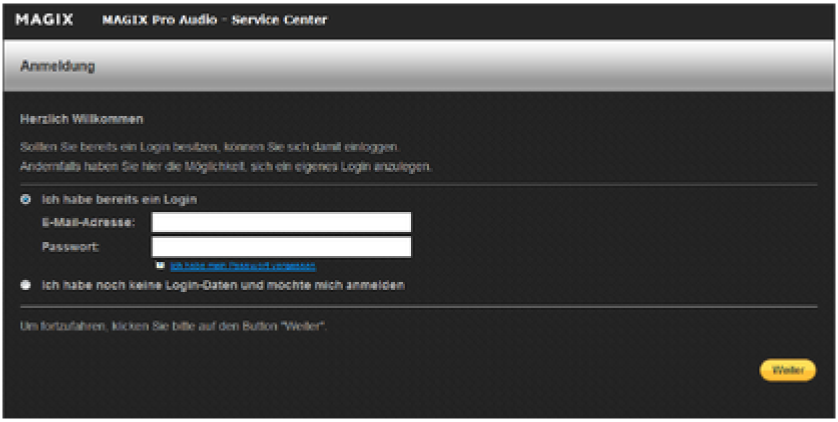
1. After starting up the program an activation dialog will appear.



2. Enter your serial number and email address and click "Register online now and activate".

3. Log in at www.magix.com in the support area. If you do not have a user account, the Service Center will open. This is where you can create a new user account and register your product.

Note: Even if you have already registered MAGIX products, you may still have to create a new login account for the MAGIX Pro Audio portal.



This completes the registration of Samplitude. You now have unlimited use of the program. You can view your registered products and corresponding serials numbers in the Service Center after login. You can also update your personal data and get the latest downloads there.

Note: Please make sure to keep your serial number in a safe place. If you lose it, you will not be able to activate the product again. Substantial changes to the hardware configuration of your system may require reactivation. You can activate the product a total of three times. If you wish to active the product more than two times, you need to contact our support team.

Use Samplitude with CodeMeter dongle

Note: The dongle activation only works with serial numbers that start with the character combination "P2". If you are using a serial number that starts with "P3", please contact support to unlock your dongle.

If you already own a dongle of a previous version or purchased a dongle through a distributor, you can use it with the current version of Samplitude:

1. In the "Program activation" dialog, select the "Use dongle" button or click "Samplitude-Dongle Activation" in the "Help" menu.

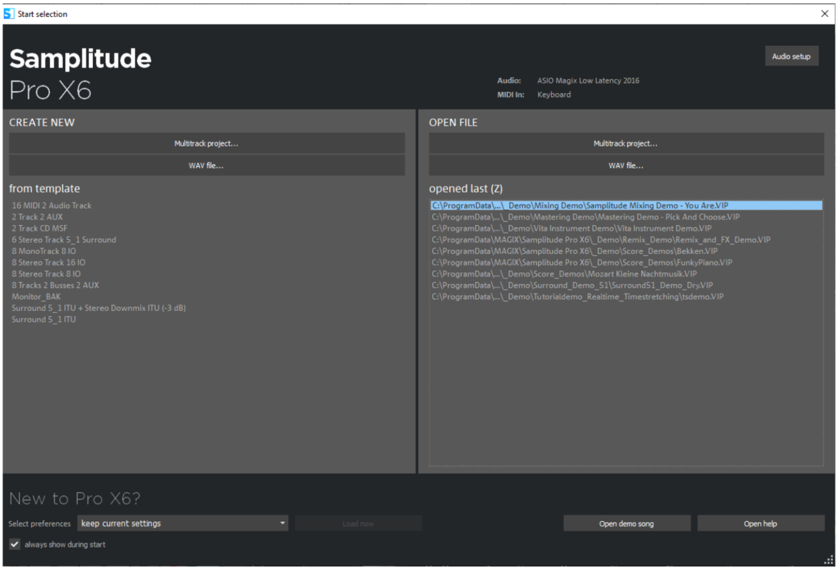
2. An information window will inform you that the use of CodeMeter is now activated. Shut the program down first.
3. Insert the dongle into a free USB port on the computer.
4. Restart the program.
5. Samplitude will now search for the "CodeMeter Runtime" and the license entries. If your dongle is already registered, Samplitude will start and you may continue with the program activation. If no license was found, an error message will appear.

If you do not own a CodeMeter dongle, you can purchase one from our sales department .

Samplitude Quick Start

This section will help you get a good start with Samplitude by introducing you to all of the most important applications necessary for an effective workflow.

When you start Samplitude you will be greeted by a "Start Selection" dialog in the program interface.



The start selection can be used to add a new multitrack project, load an existing multitrack project, load an audio file and access the options for recording an audio file (recording a "Wave").

You can also use it to display a demo project, open the Help section, load the most recently opened projects or to start the intro video.

Clicking the corresponding check box ensures that the start dialog will always open when the program is started.

Initial Navigation In The Virtual Project (VIP)

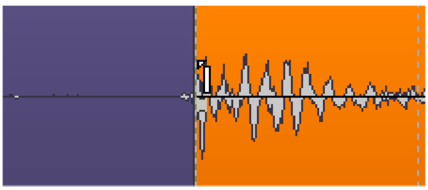
- To start playback, press the space bar.
- To end playback, press the space bar, the pause button, or the "0" on the keyboard.
- To set the playback marker, click on the desired position in the timeline.
- To move the playback marker, use the left and right arrow keys.

- To jump to the most recent position, press the back space key
- To move the VIP under the playback marker, use the keyboard shortcut "Alt + Arrow left"/"Alt + Arrow right".
- To create a range, use the mouse to draw in the grid list.
- To move a range, move it with the mouse while holding down the Shift key.
- To increase the size of a range, drag the end of the range using the mouse.
- To reactivate a deactivated range, use the shortcut keys "Shift + back space".
- To move an object, click on the lower half and drag it into its new position.
- To jump to the object edges with the playback marker, select the shortcut keys "Ctrl + Q"/"Ctrl + W".
- To create markers at the playback marker position, select "Shift + 1...0" using the number keys along the top of your keyboard.
- To jump to a marker position with the playback marker, select the corresponding number "1...0" using the number keys at the top of your keyboard.

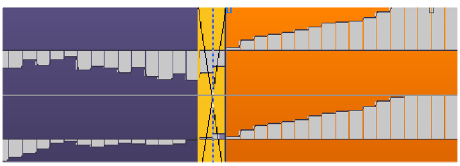
Detailed explanations of additional navigation functions can be found in the chapters "Working in the project window", "Methods for working with objects (view page 158)" and in the menu reference, help file, or complete documentation "manual.pdf" under "Edit" > "Range (view page 655)". A complete overview of all keyboard shortcuts can be found in the chapter "Preset keyboard shortcuts".

Cutting And Editing Objects

Position the playback marker at the cut position. Now click on the object to be cut and press the "T" key on the keyboard.



If "Auto Crossfade" mode is activated ("Edit" > "Crossfade" > "Auto Crossfade active"), Samplitude will create a crossfade between the two newly created objects.



Each object has its own object editor that you can use for detailed object editing. The object editor can be opened by double-clicking on the object.

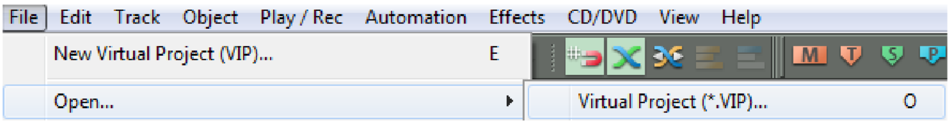
Detailed information on this can be found in these chapters: "Object-oriented audio editing (view page 64)", "Object editor (view page 146)","Working with objects (view page 158)" and "MIDI in Samplitude (view page 324)".

Arrangement Of A Song (Example)

This workshop will show you how to edit and arrange audio material using Samplitude. It will introduce you to all of the basic functions for editing process.

Load Project

Go to "File" > "Open" > "Virtual Project" to open the project file "WS1.VIP" which is required for this workshop.



You'll find them in the folder "ProgramData" > "MAGIX" > "Samplitude" > "_Demo" > "Workshop_Edit_Arrange" > "WS1.VIP".

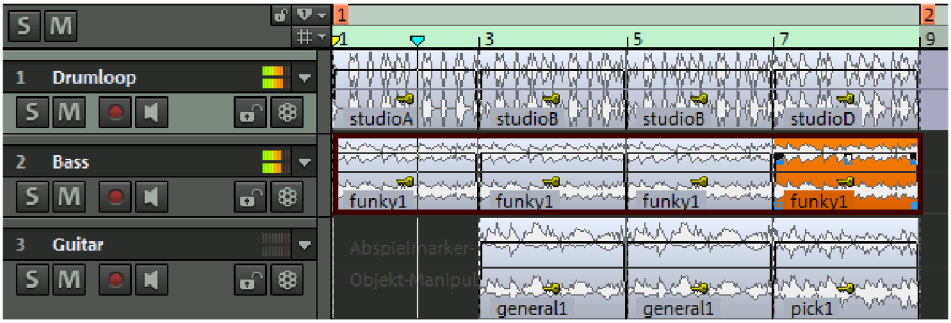
Play Project

Press the play button and listen to the source audio material as a loop. To do this, press the "Loop" button on the transport console and highlight the grid/marker bar by double-clicking between marker 1 and 2.   Playback can also be started using the space bar.

Copy Object

As you can hear, the bass only plays for two bars. Activate the bass object by clicking on the lower half of the "funky1" object. Its color changes to orange.

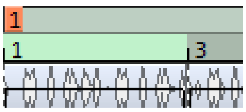
Now copy the object with "Object" > "Edit" > "Duplicate Objects". You can also use the keyboard shortcut "Ctrl + D" or drag & drop while holding down the Ctrl key. Now your bass should play until the end of the eighth beat.



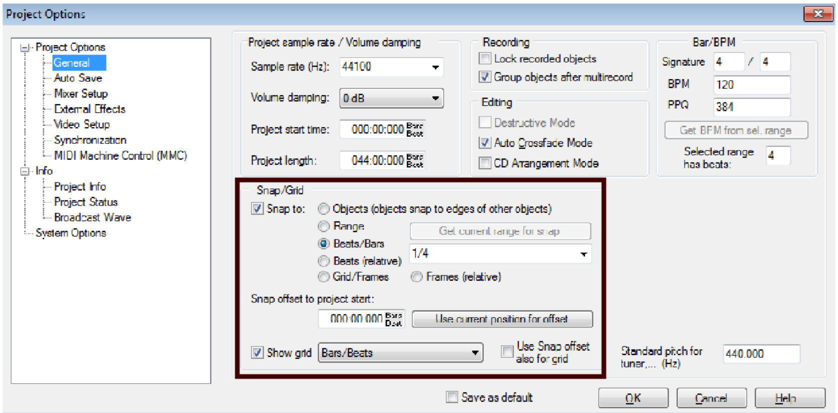
Current status WS2.VIP

Copy And Paste A Range

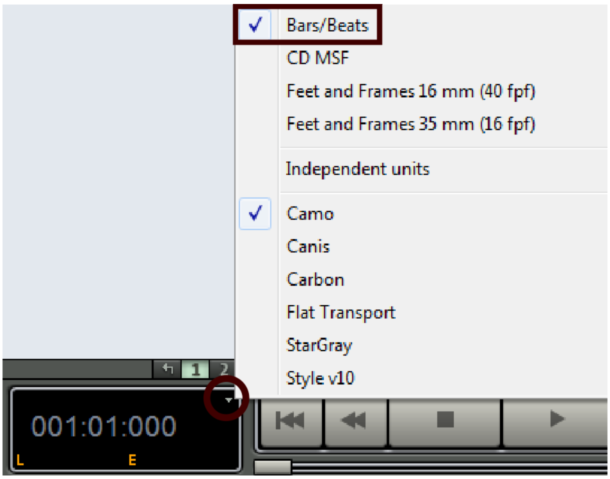
The next thing to do is duplicate both of the first bars. To do this, draw out a new range to the start of the third beat in the grid and marker bar.



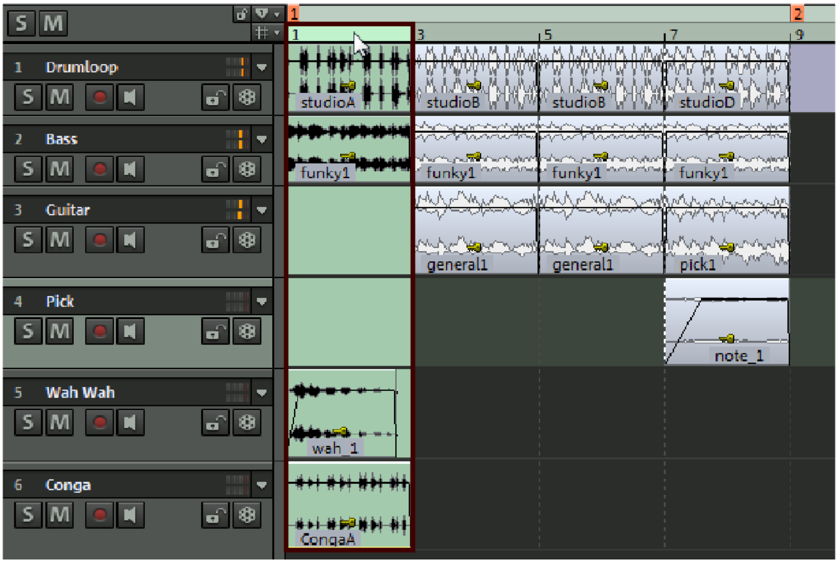
Here the grid and snap function should be switched on in the project options (shortcut: I) and "Snap to" and "Beats/Bars" should be selected.



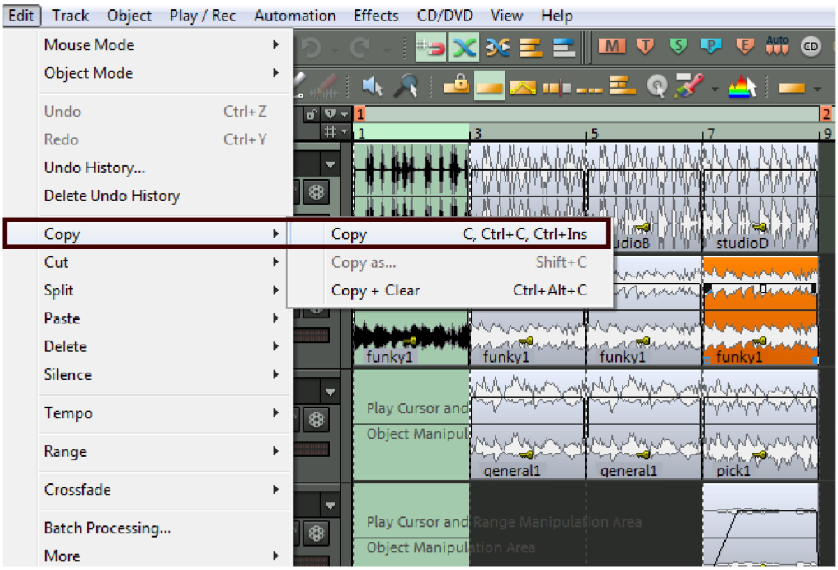
Bars/Beats should also be selected in the transport window beside the play position display.



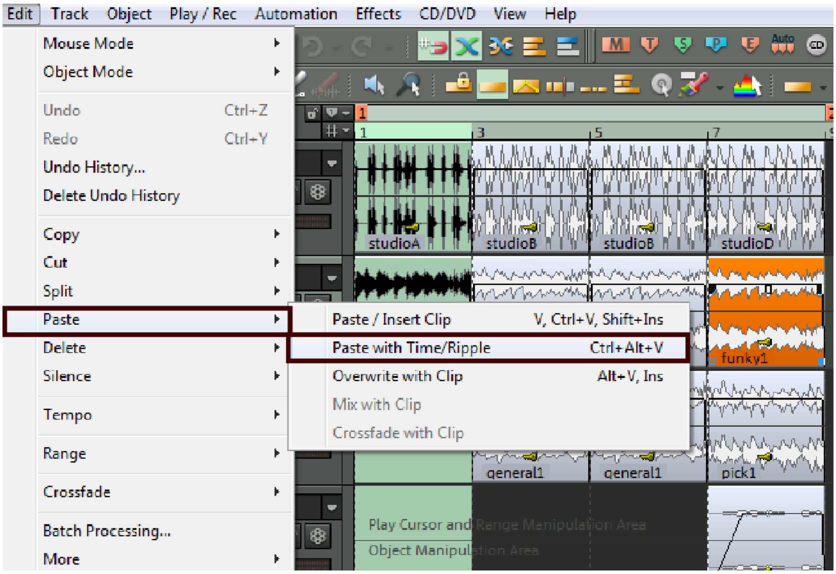
Now double-click in the selected area. The first double-click selects the range in the currently selected track. By double-clicking again all tracks in this range will be highlighted.



Now go to "Edit" > "Copy" > "Copy" (Ctrl + C) to save the selected range on the clipboard

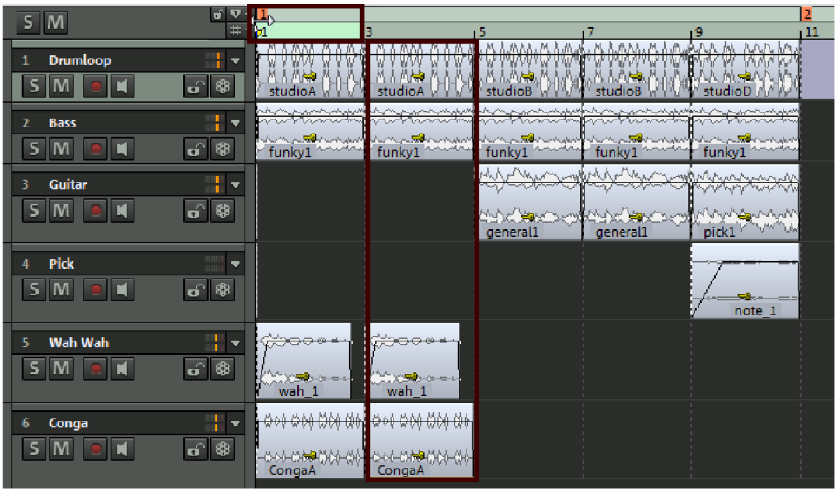


and then use "Edit" > "Paste" > "Paste with Time/Ripple" (Ctrl + Alt + V) to insert it on the right.



The part of the arrangement that comes after it will be moved two bars forward.

Now use the mouse to move marker "1" to the beginning of the project.



Current status WS3.VIP

Duplicate the range between the seventh and ninth bars in the same way.

Current status WS4.VIP

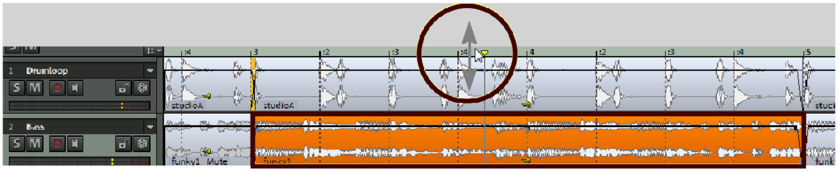
Now if you double-click on the grid and marker bar between marker 1 and marker 2 while the "Loop" button on the transport console is activated, you can play back your new arrangement by pressing the space bar.

Mute Objects

To make the intro sound a bit more interesting, we'll introduce the bass later on. To do this we'll mute the first bass object using Ctrl + M.

Zooming In And Out Of The Project

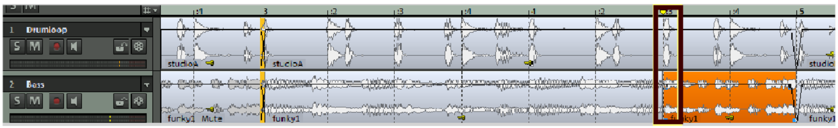
Now click on the object beside it. For the following editing function we'll increase the display size of the activated object. You can quickly and easily zoom in by left-clicking on the grid/marker bar at the position of the current object, holding down the mouse button and dragging it down vertically.



This way you can work very precisely in the waveform display. If you press the Ctrl key before letting go, you will return to the original zoom level.

Split Object

Now move the cursor to the position 4:03. At this position in the bass track you can split the object with the shortcut key "T".

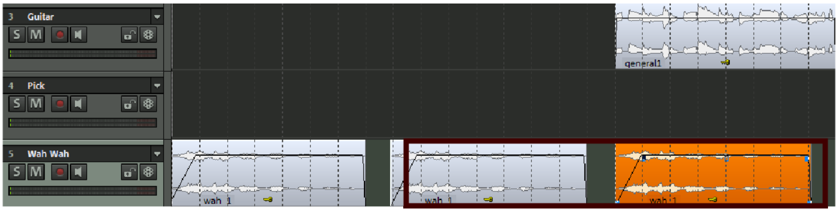


The cut will create two objects. Zoom out of the detailed display slightly (to do this you can drag on the right end of the scroll bar) and mute the longer object in front by going to "Object" > "Edit" > "Mute Objects" or using the keyboard shortcut Ctrl + M.

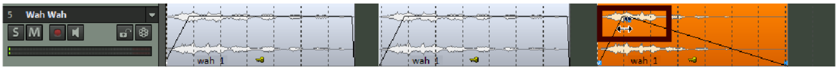
Fading Objects In And Out

The remaining bass part ends the intro. You can add a fade in by dragging the top left handle of the bass object to the right.

At the beginning of the chorus create a copy of the Wah guitar by dragging the mouse while holding down the Ctrl button.



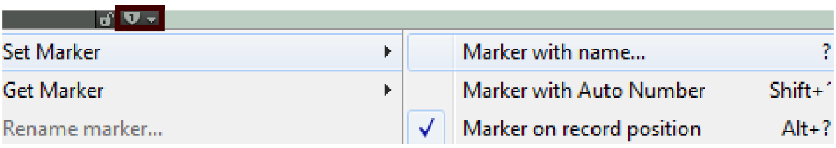
We want to let this fade out right after it is played. We can do this by dragging the object handle (small square) on the upper right edge of the object all the way to the object start on the left. This is how you edit the fade out phase of the object.



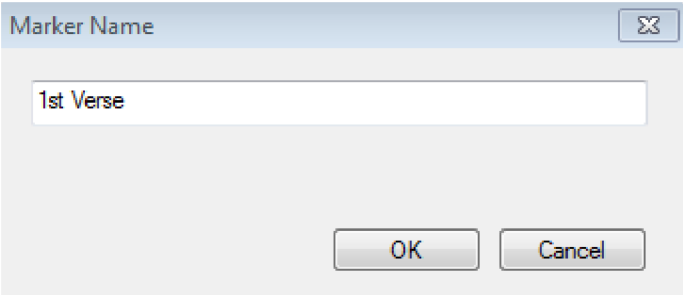
Setting And Naming Markers

Now we want to set new markers. First we have to delete all of the old markers by clicking on the marker button beside the grid and marker bar and selecting "Delete all Markers".

Now move the playback marker to the beginning of the arrangement by pressing the Home key on your keyboard. Right-click on the marker button again and go to "Set Marker" > "Marker with name...".(Keyboard shortcut: "?").

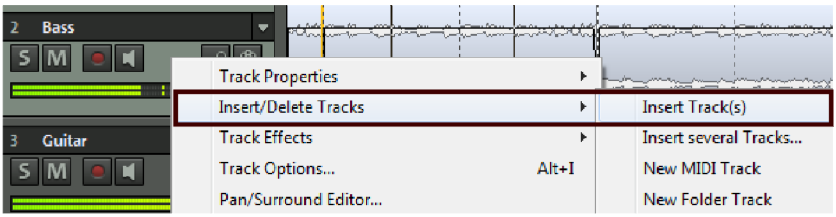


Give the name "Intro" to the marker at the beginning of the song and do the same for the marker you want to create at the beginning of the first verse at the song position 005:01:000 by giving it the name "1st Verse".



Creating And Naming A Track

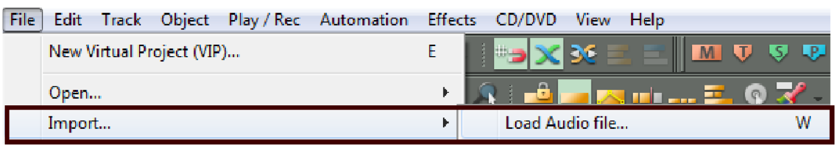
To ease the intro into the verse a bit more smoothly, insert an additional bass part and add it on its own track. To do this, click on the existing bass track, then right-click on the track header and select "Insert/Delete Tracks" > "Insert Track(s)" from the context menu.



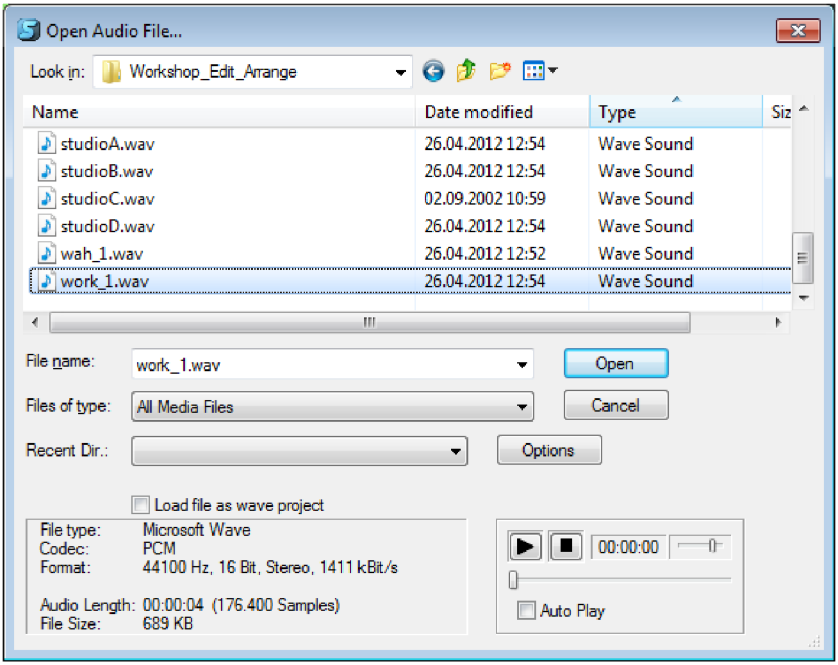
Double-click on the name field of the new track and enter the track name "Bass2". Confirm this by pressing Enter.

Loading Audio File And Adding Them To The Arrangement

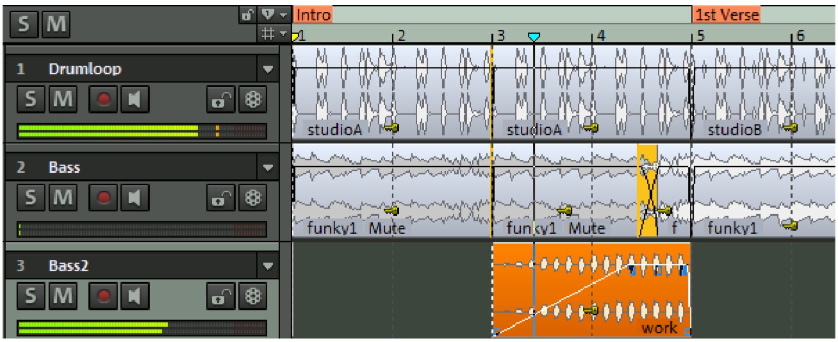
Now move the playback marker to the bar position 003:01:000 and open the audio import dialog through "File" > "Import" > "Load Audio file" (keyboard shortcut: "W").



Click on the file "work_1.wav" in the "Workshop_Edit_Arrange" folder.



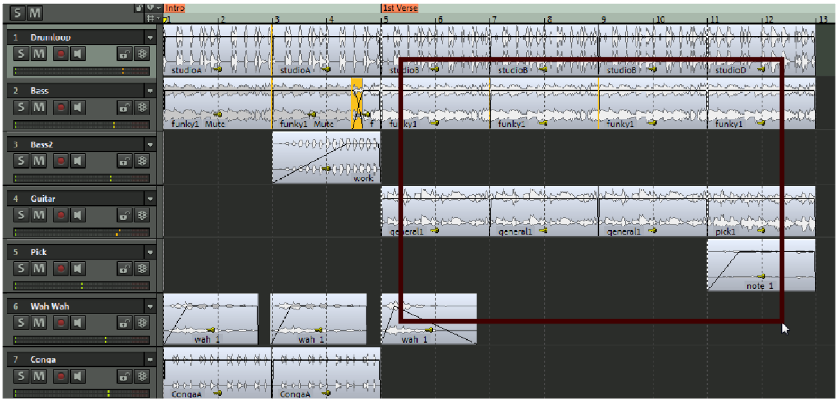
When you click on "Open" the bass object will appear on the new track at the position of the playback marker. Now create a "fade in" for the inserted object and listen to the result.



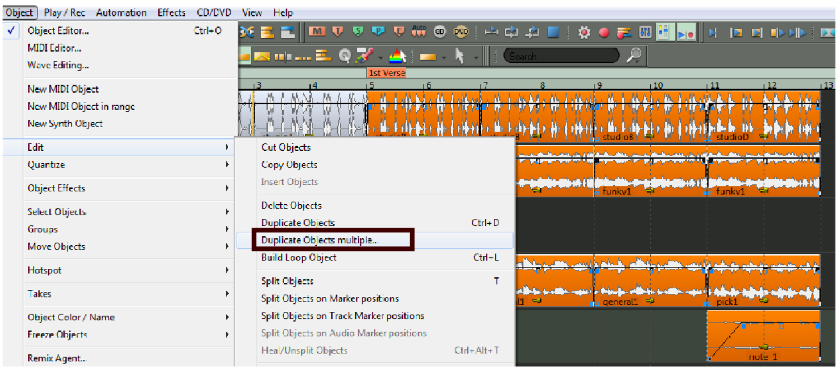
Current status WS5.VIP

Multiple object copying

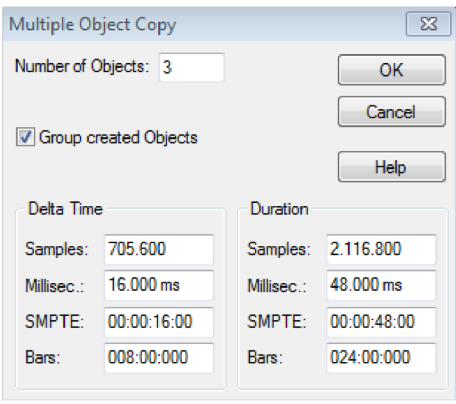
To expand the song structure, copy the objects in the first six tracks of the first verse. Place the mouse cursor over an empty area of the sixth track and drag the mouse up and up to open the object lasso. All objects in the lasso are selected when the mouse button is released.



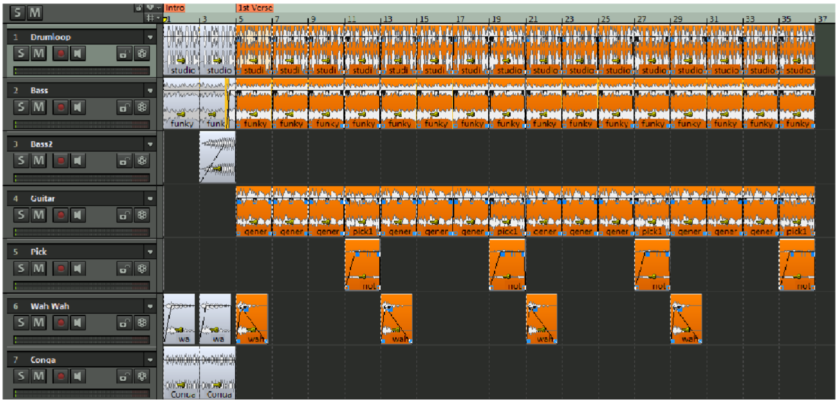
You can duplicate the objects you selected by going to "Object" > "Edit" > "Duplicate Objects multiple...".



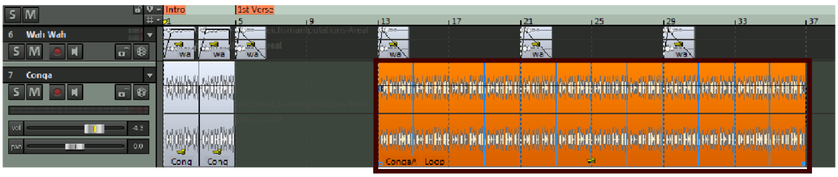
In the dialog that follows, enter "3" as the number of objects.



The newly created duplicates now link up with the existing objects. To display an overview of all the objects, you can use the keyboard shortcut "Ctrl + Alt + Up Arrow".



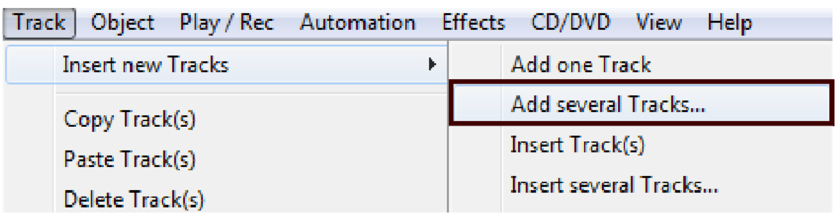
Now, copy the conga object from track 7 to the start of bar 13. Right-click on the object and select "Create looped object" (or press Ctrl+L). Drag the right handle to the right to create a loop object which you can set to any length you want. Lengthen the conga loop this way until you reach bar 37.



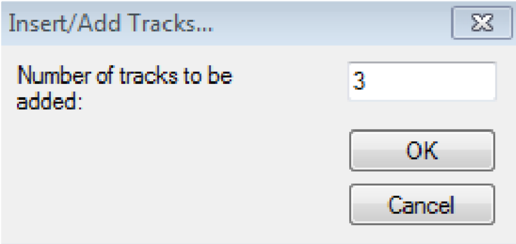
Current status WS6.VIP

Adding Several Tracks

Now create three new tracks at the end of the arrangement by going to the "Track" menu > "Insert new Tracks" > "Add several Tracks"



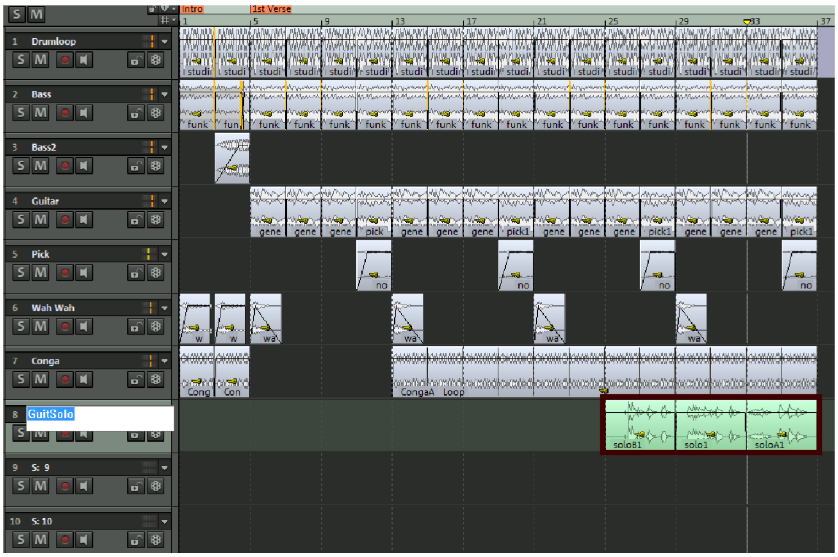
and entering "3" as the number of tracks you want to insert.



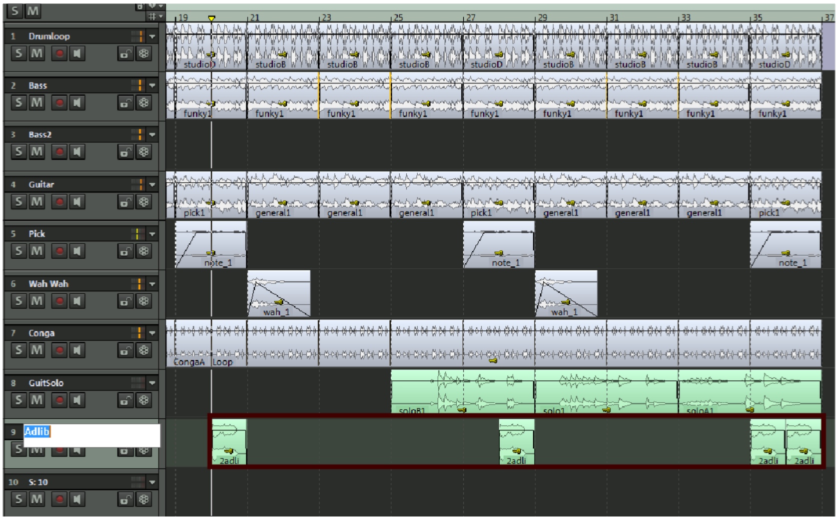
To continue to view all objects in the overview vertically, click on the "Overview Mode" button on the lower toolbar.

Positioning Audio Files In The Arrangement

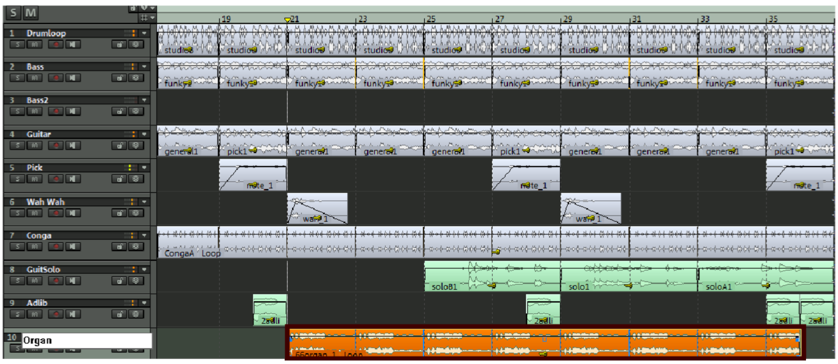
Using the keyboard shortcut "W", first load the audio files "soloB1", "solo1" and "soloA1" one after the other into track 8 at beat positions 25, 29 and 33. Change the track name to "GuitSolo".



Similarly you can change the name of track 9 to "AdLib", load the audio file "2adlib1.wav" to the start positions of the bars 20 and move copies of the object to the start positions of the bars 28, 35, and 36.

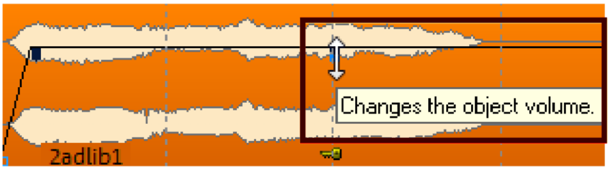


Finally, name track 10 "Organ" and load the audio file "66organ_1.wav". Now build a looped object in track 10 with Ctrl + L and place the loop on bars 21 to 36 of the VIP project.



Adjusting Object Volume

The final step is to adjust the volume of the vocals, guitars, and keyboard using the upper middle handles of the respective object



or by setting the volume of the corresponding track with the volume fader in the track header or the track editor.

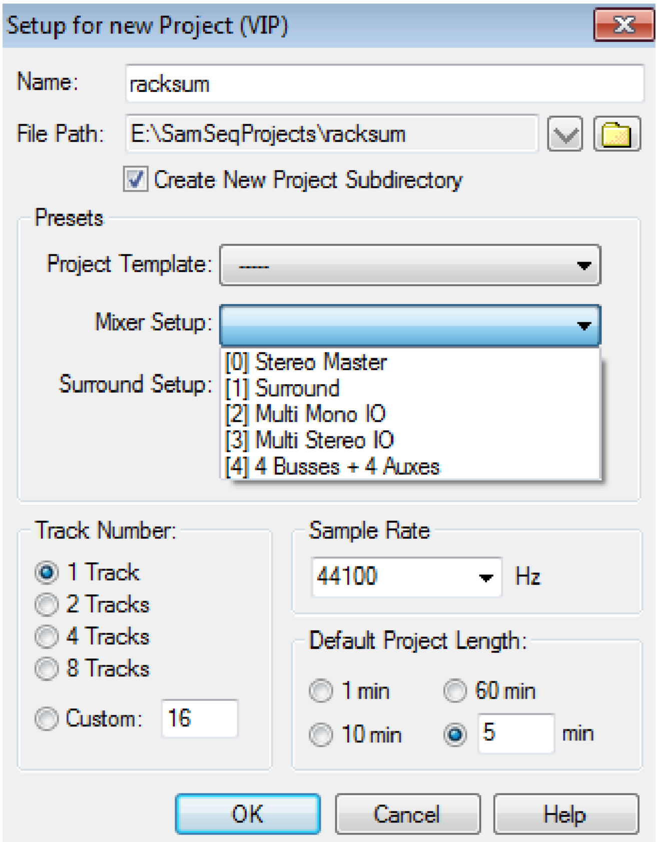
Try experimenting with the arrangement even further by extending the chorus, middle eight and outro or adding effects and automation.

Audio Recording

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

Creating A VIP

First of all, create a new project by clicking on the "New Multitrack Project (VIP)" button in the program's Start Wizard menu. If you already have Samplitude open, go to "File" > "New Virtual Project (VIP)...".



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

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

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

Preparing To Record

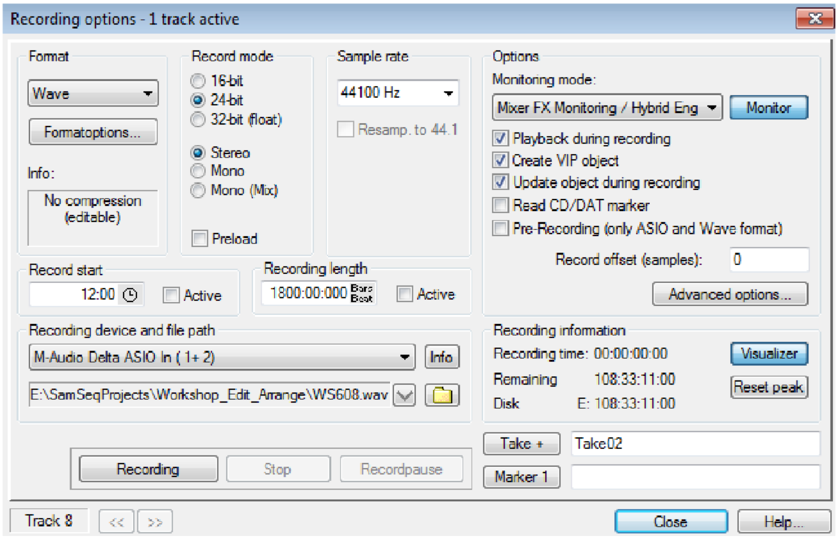
To record from your internal CD/DVD drive or from an external sound source using a cable/microphone connection, it is important that your sound card is connected to the sound source. You can check this by switching on the track that you want to record. Click the record button in the track header to do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

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

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

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

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



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

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

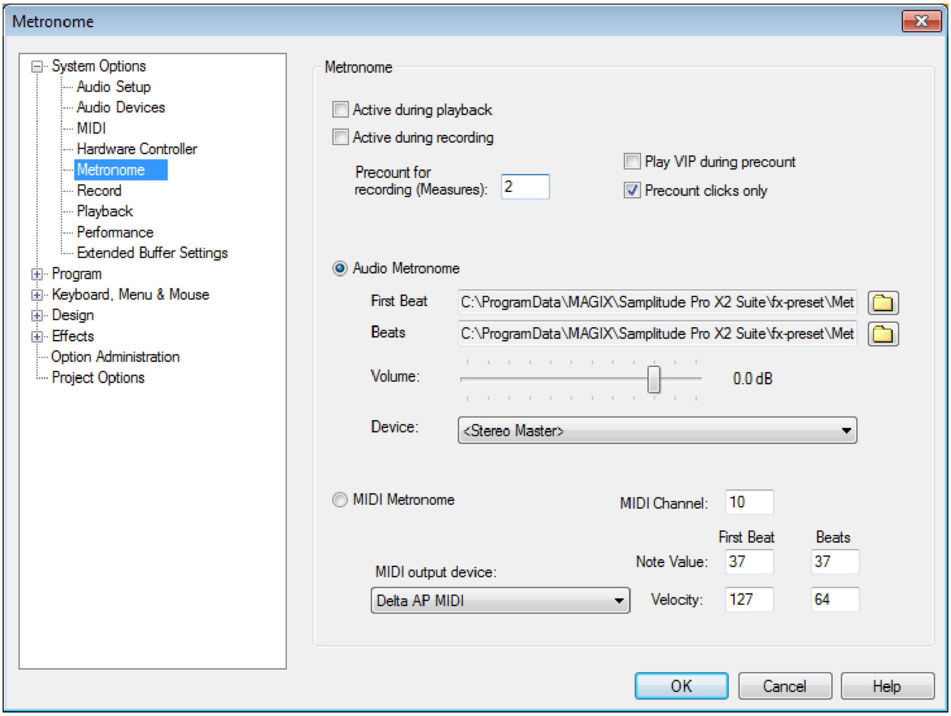
Now specify if the track should be recorded in mono or stereo. If you are recording from a sound source such as CD or DVD, it is best to use a stereo recording mode. For single instruments or vocals, it is better to use a mono recording mode.

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

Detailed information about the recording options is available in "Play/Rec" > "Recording Options (view page 732)"

Metronome Options

You can activate the metronome through "File" > "Program Preferences" > "Metronome Options...". The metronome can be set as an audio metronome or as a MIDI metronome.



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

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

Precount For Recording (Measures): Here you can specify the number of beats that the metronome will count in before the recording starts. You can also make the VIP play back during the precount by activating the option "Play VIP During Precount". If you activate "Precount Clicks Only", the metronome will stop when the recording starts.

The "Punch Recording" button in the toolbar blinks during precounting and the "Record" button blinks in the transport console (view page 90). When recording begins, both buttons remain active until the end of the respective recording run.

Detailed information about different punch functions is provided in the "manual.pdf" documentation, in the help file, and under "Play/Rec" > "Record Mode/Punch In" > "Punch Marker Mode (view page 730)".

Audio Metronome

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

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

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

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

MIDI Metronome

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

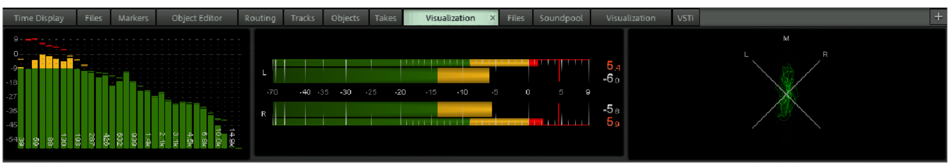
MIDI Channel: Here you can set the MIDI channel through which the MIDI commands are sent.

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

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

Level Adjustment

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



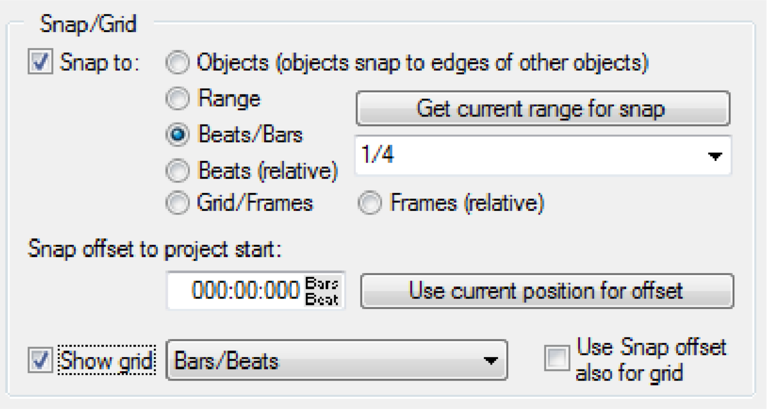
If there is no visual display of levels in the peak meter, click the "Mon" button on the transport console.

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

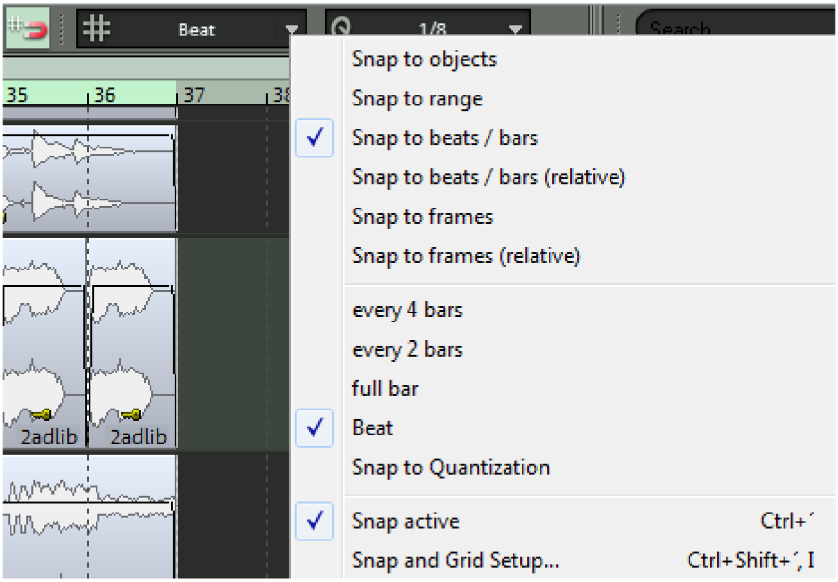
Snap And Grid Setup

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

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

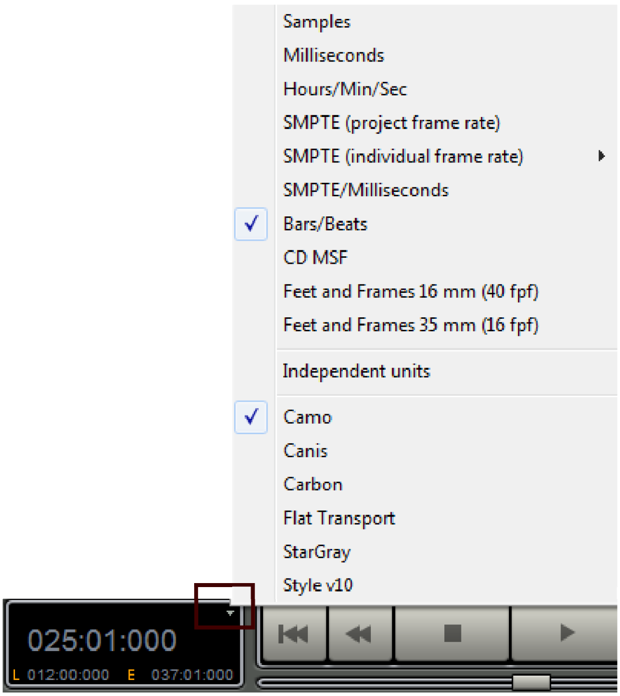


Now you can see the grid lines in the VIP.



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

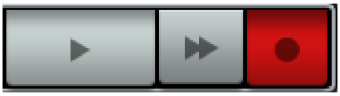
Step 2: Switch the display in the transport window (view page 90) to "Bars/Beats" using the start display area.



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

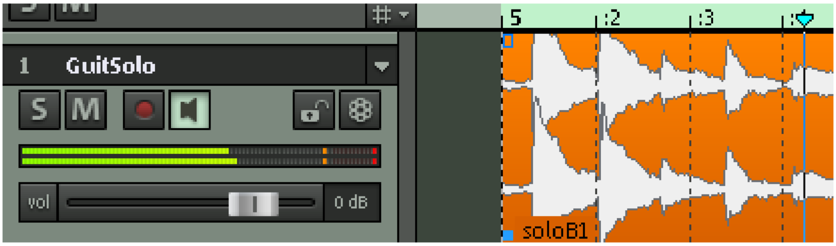
Starting To Record

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



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

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



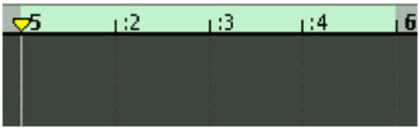
Now activate the next track by clicking on the record button in the second track.

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

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

Loop Recordings

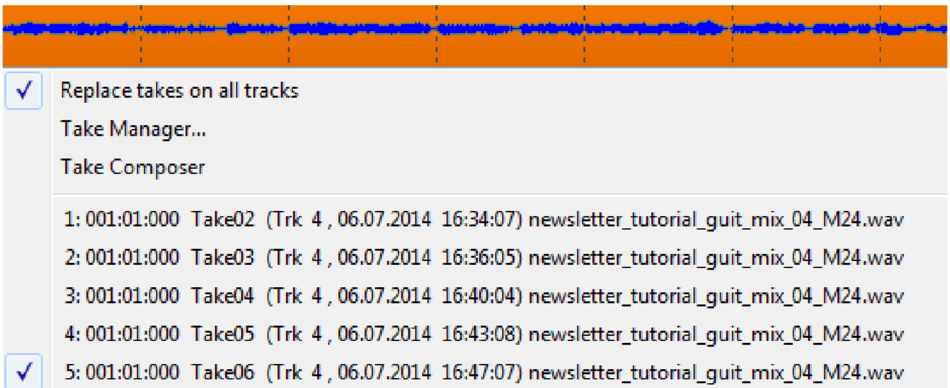
Step 1: If you want to record in a loop, first select the range in the arranger that you want to record.



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

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

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

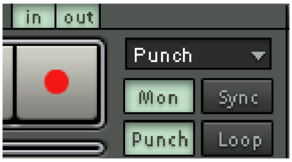


The Take manager (view page 189) is useful for organizing and editing recording runs. Use it together with the Take composer (view page 192) to produce the perfect take.

Punch Recording With Markers

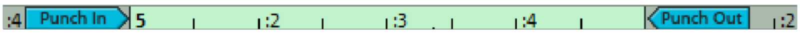
If you know the exact range that you want to record in, punch recording using markers is the best option.

Step 1: To do this, select the area with the mouse where you want to use punch recording, and then activate the "In" (sets the punch-in marker) and "Out" (sets the punch-out marker) buttons in the transport console (view page 90).



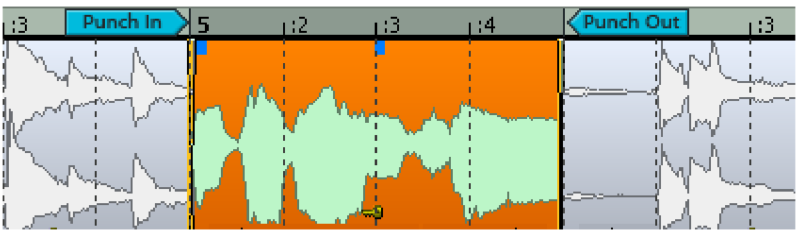
Step 2: Now you can set the playback marker in position.

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



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

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

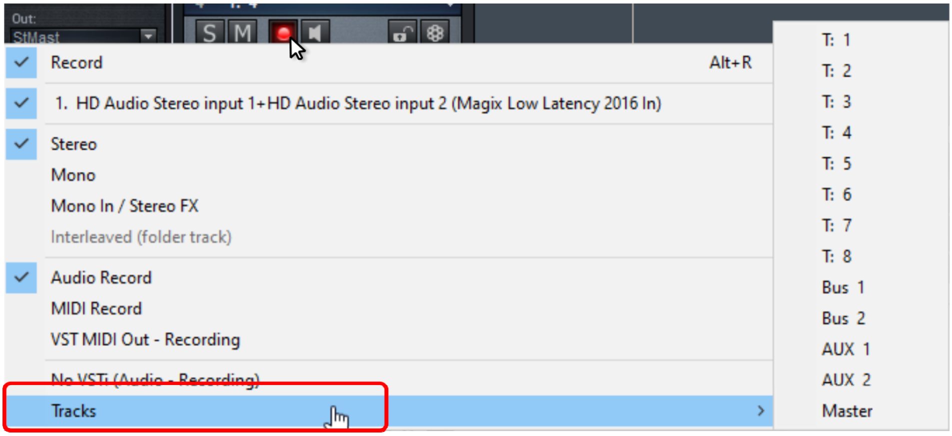


Detailed information about different punch functions is provided in the "manual.pdf" documentation, in the help file, and under "Play/Rec“ > "Record Mode/Punch In“ > "Punch Marker Mode (view page 730)".

Recording track output

It is also possible to record the output of a track, bus, or master to a new track.

To select the output of a track as the source for audio recording, right-click the Record button in the Track Header, 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 bottom, in **Tracks**, you can select one of the tracks or masters of the project as the recording source.

This allows you, for example, to quickly output an intermediate state of the project by recording the output of the master to a new track without having to worry too much about the render settings.

MIDI Recording

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

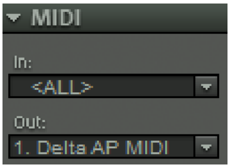
In a Samplitude, MIDI data is recorded similarly to audio data. You create an object for each recording, which overlaps existing objects.

Preparing A MIDI Recording

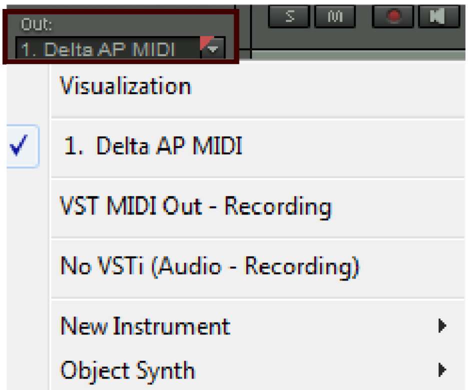
If the "MIDI" button is activated in the track editor, the MIDI section opens to prepare the track for MIDI recording.

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

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



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



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

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

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

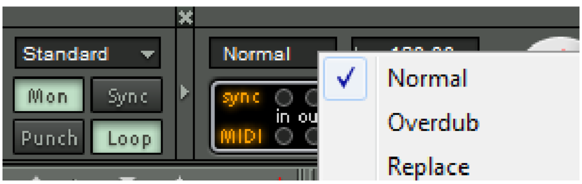
Make sure that your MIDI synthesizer is sending to the same MIDI channel that you have selected in the "channel out" slot in the track editor. To be certain, set the Out channel to "All". Now you should here all of the synthesizers no matter what channel they are on.

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

MIDI Record Modes

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

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



Normal: This recording mode is the equivalent of a normal audio recording. This means that a new MIDI object is created over the existing object for each recording. The old object remains, but it can be covered partially or completely by the most recently recorded one (both visually and acoustically). This way, you can record multiple takes of a passage and then compare them in the take manager (view page 189) later on.

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

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

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

Now you can now start the MIDI recording using the shortcut key "R". After ending recording with the "Stop" button, you will be asked whether you want to keep or delete the recording. After confirming this with "OK", your newly recorded material will appear as a "MIDI take" in the VIP. In case you have recorded several takes in normal mode for the same selected range, you can select and play back the individual MIDI takes in the Take manager (view page 189).

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

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

CD Mastering

CDs can be burned directly from within the project without having to export the project as a wave file beforehand. Burning requires that a blank disc be inserted into the drive and a valid TOC (Table of Contents) exists. Place at least one CD track marker and a CD end marker to do this.

Setting A CD Track Marker

Move the playback marker to the position where the CD track marker will be. The marker menu is located above the track headers in the VIP. Open it and select "Set

CD Track Index". Repeat this process until all markers that you want to set are in place.

Now move the playback marker to the position where the CD will end. Open the marker menu again and select "Set CD End Marker".

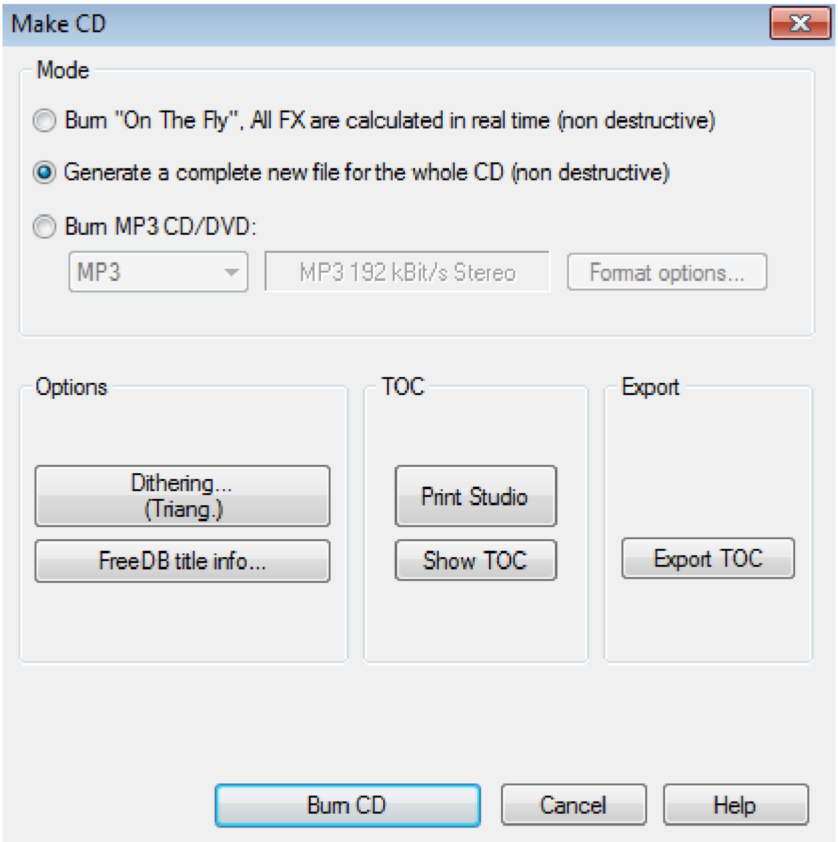
Note: The burning process only starts from the first CD Track Marker.

Burn CD

Now play the project again and at keep an eye on the Samplitude status bar to see how much CPU is required to play the project. This is important in order to estimate the speed that the project can be burned to audio CD.

Now click the button with the CD symbol. Alternately, select the "Make CD..." option in the "CD/DVD" menu. The CD burning dialog now opens.

Make CD



Mode: Select whether you want to burn your project directly "on-the-fly" (without prior rendering) or whether Samplitude should create a new file beforehand (bouncing).

Click "**Burn CD**".

CDR write settings: Enter the desired burn speed here.

If to wish to use CD text, open the "**CD-Text/MP3 ID editor**" by clicking "CD-Text settings" and entering the desired CD text for your tracks.

Click on the "**Write**" button to begin burning. During burning, the play cursor runs through the project to provide a progress display.

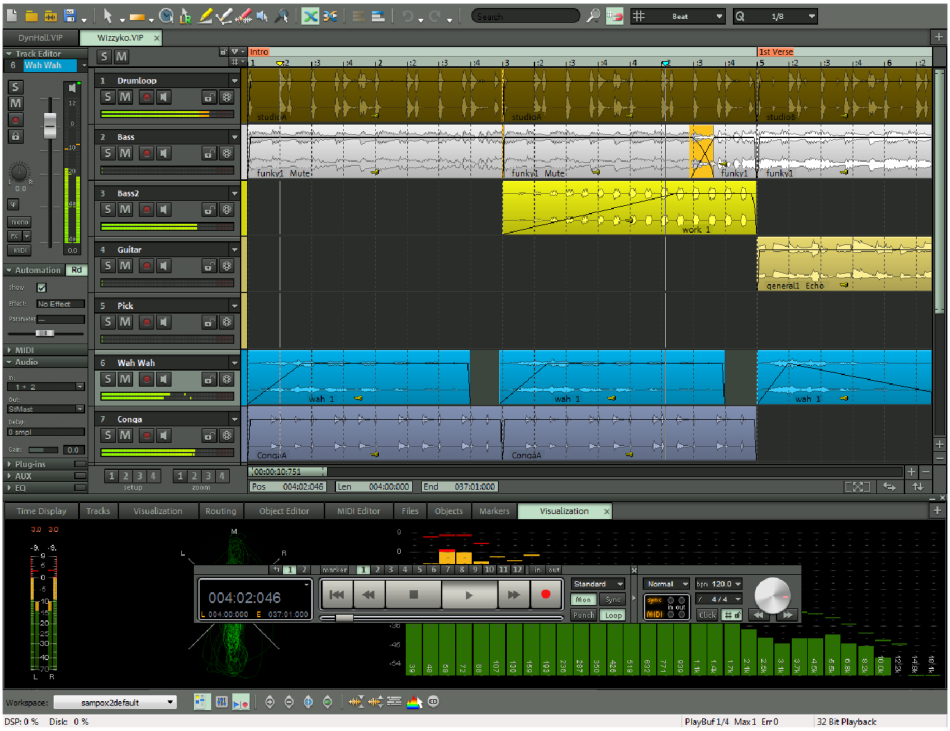
After burning, a message appears indicating that the process is complete. Confirm this by clicking "**OK**".

Detailed information about CD mastering is available in the menu reference under "CD/DVD". (view page 982)

Samplitude Fundamentals - Basic Terms

VIP

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

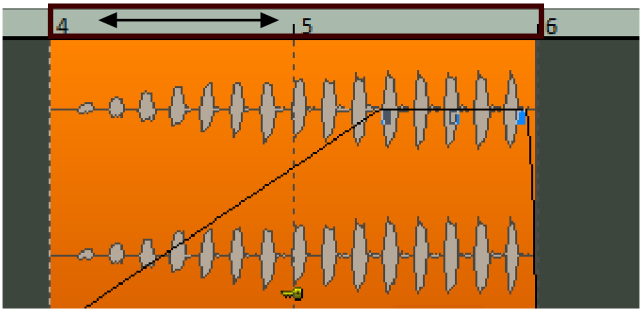


The file extension .vip is applied to all virtual projects in Samplitude.

Snap

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

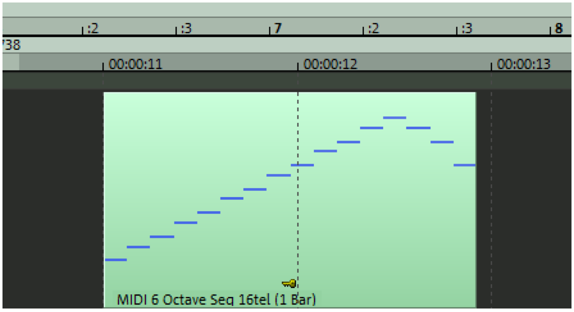
For example, the snap function is used to move objects and position them precisely on snap points in the grid.



Grid toolbar with the snap setting "Beat".

Grid

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

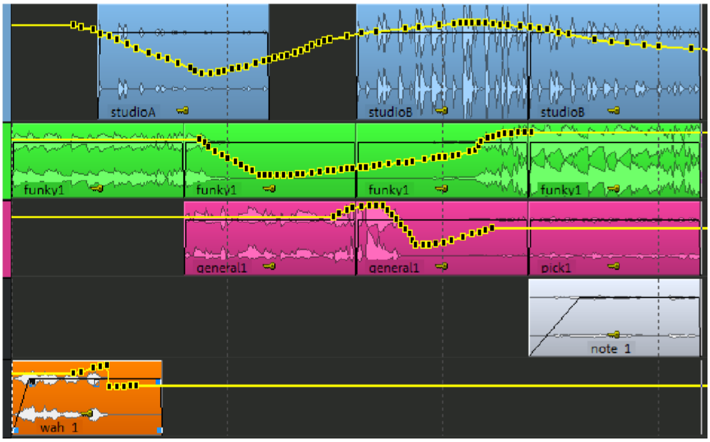


VIP with two grid bars - Grid setup based on the time unit of the lower grid toolbar.

More information about snap and grid settings can be found in the chapter „Screen Elements > Toolbars - Overview > Grid toolbar/snap buttons“ (view page 99).

Arranger

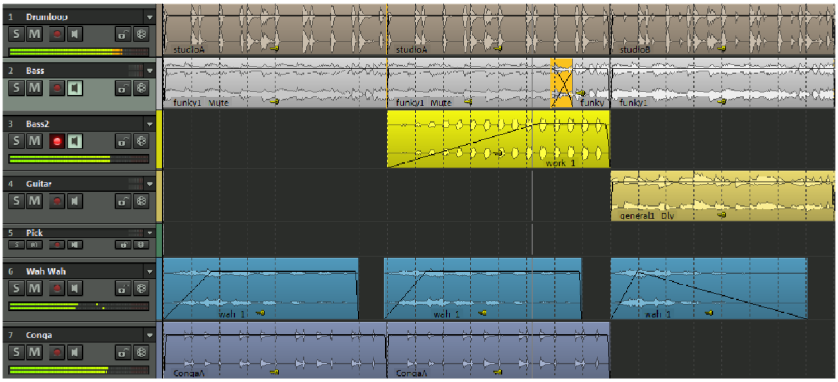
The arranger represents the larger portion of the project window. It's where you manage the temporal arrangement of objects on tracks.



Detailed information about how to work with the arranger can be found in the chapters „Working in the Project Window“ und „Working with Objects“ (view page 158).

Tracks

Multiple simultaneously played or recorded audio or MIDI objects are placed on top of each other on different tracks. Each track corresponds to a channel on the mixer (view page 203). You can use this channel to control volume, apply effects, mute or solo all of the objects in the track.



Section

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

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

Several Ranges

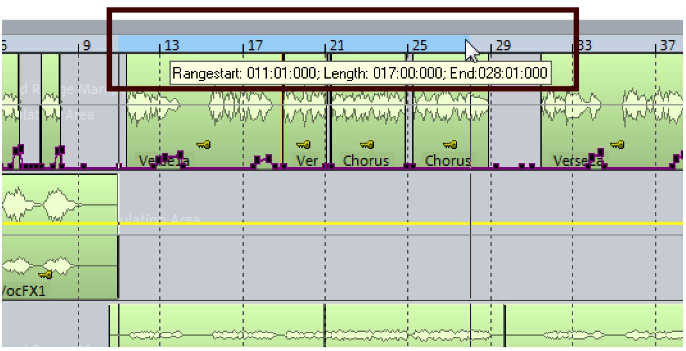
If you display your project in several ranges (menu "Edit" > "Range" . "Split range"), only one range can be active at a time. Activate a part by clicking it or clicking its controls. By clicking both double arrow buttons at the left lower border of a range, a corresponding context menu will appear that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

Additional information about sections is available in "View > Sections (view page 1032)" and "Edit > Range > Split Range (view page 659)".

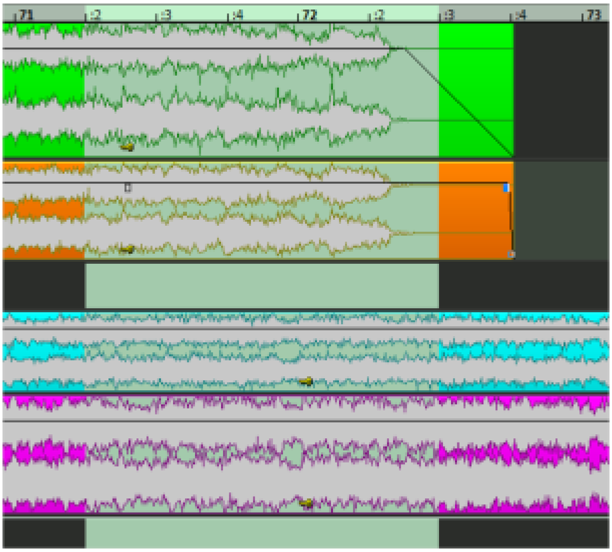
Ranges

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

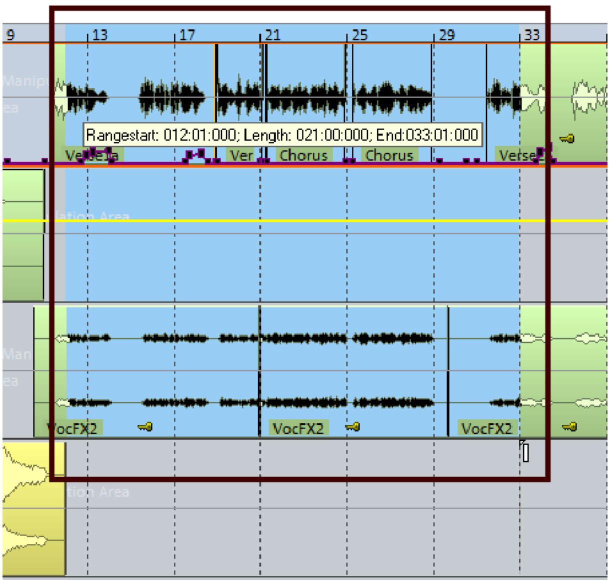
Playback ranges: Playback ranges can be selected above the first track in the grid bar via drag & drop.



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



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



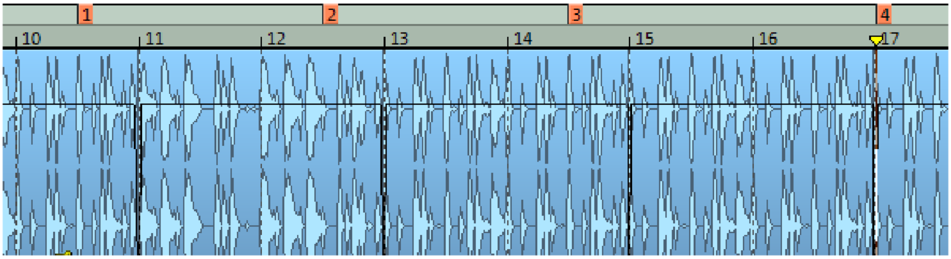
You can also use the grid feature when creating ranges. Ranges can also be saved for later use ("Edit" > "Range" > "Store Range/Get Range")

See Working with ranges (view page 130) for further details.

Detailed information about working with ranges can be found in the "Working in the Project Window -> Working with Ranges (view page 130)" chapter.

Markers

Use markers to select important sections in your arrangement.



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

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

Keyboard shortcut:

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

Get marker: Number key

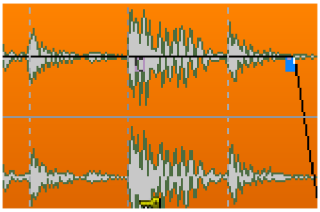
In addition to the standard markers, there are other markers for special functions: Audio markers, markers for CD burning functions, tempo markers, bar markers and grid position markers.

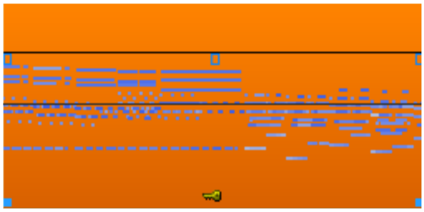
See the chapter on "Working with Markers" (view page 134) to find out more about markers.

Detailed information about working with markers can be found in the "Working in the Project Window" -> "Working with Markers (view page 134)" chapter.

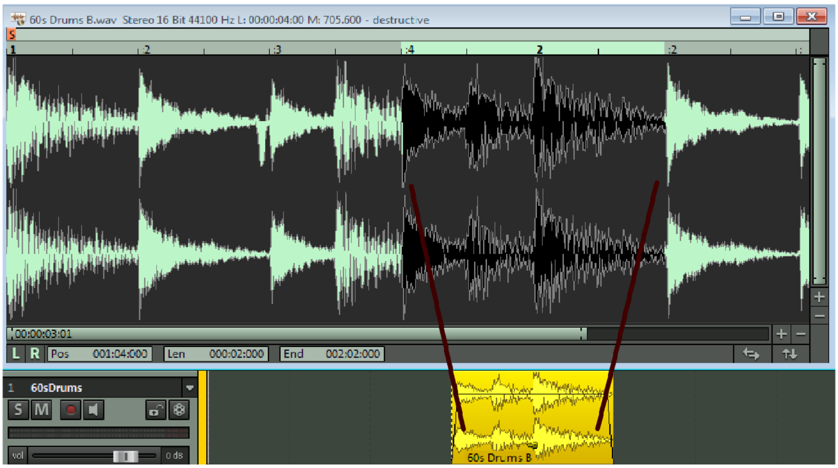
Objects

The object is a unit in the Samplitude arranger window. Objects are created when you record audio or MIDI data as well as when audio or MIDI files are imported into Samplitude.





Audio objects are references to underlying audio files. Typically, people say that an audio file is being referenced by an object. These are essentially playback instructions that determine what parts of what audio files should be played in the arrangement and when they should be played. The starting point of an object is assigned to a specific point in an audio file. The length of the object determines the length of the part of the audio file.



When you edit objects (by adding transitions, loudness adjustments, effects, cuts, etc.), new instructions are defined which then get carried out in real time during each playback. The actual audio file is not changed ("non-destructive editing") but the adjustments to the settings are saved permanently.

Since objects are merely playback instructions and only refer to audio material, they can be moved, copied and deleted without affecting the audio files themselves.

The audio material being referred to by the audio object is displayed as a wave form.

MIDI objects to do not reference MIDI files. They do get created when you import MIDI files, but the MIDI data is saved entirely to the object and is therefore directly linked to it.

Object-oriented Audio Editing

Object-oriented editing describes a method which enables the various ways of editing any selected object, independently of other mixer, track or automation settings. This results in a high level of flexibility, even on the object level. You can edit audio material faster, add various effects to individual objects and assign object-specific AUX components. The original audio files remain unchanged.

The advantage of realtime editing is that the objects do not have to be rendered or bounced. In contrast to editing with track effects, the calculation of the object effects only takes place when the object is played back. This lets you minimize the CPU load on your computer.

Object Editor

The object editor makes audio object editing more efficient by providing a clear overview of all the editing functions in a display that is similar to a channel strip. This means that with the object editor Samplitude offers a "realtime channel strip" for each object.

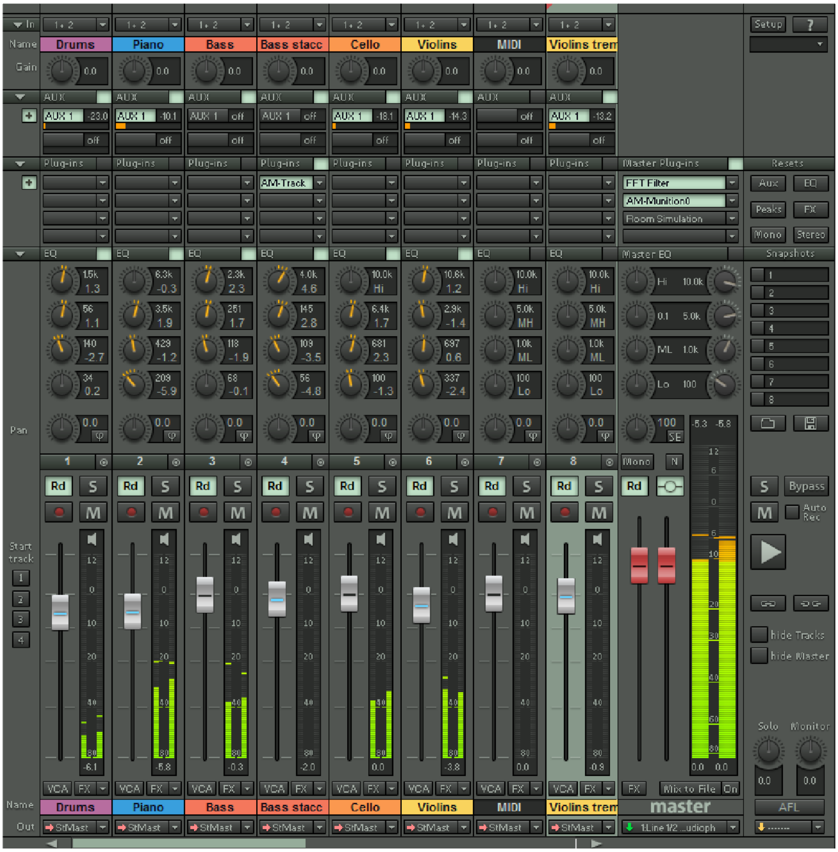
If you cut an existing object in two (keyboard shortcut: T), you will have two objects which each have their own object editor. The objects can also be cut again to form even more objects. This allows you to quickly and directly apply all of the editing options mentioned above to any of the individual audio objects.

The editing possibilities offered by this feature often allow you to avoid further editing steps, such as track automation, etc.

Detailed information about the object editor can be found in the chapter "Object Editor (view page 146)".

Note: There is an object editor for MIDI objects as well. Detailed information can be found in the chapter "MIDI in Samplitude -> Working with the MIDI object editor (view page 326)"

Mixer



You can use the mixer (keyboard shortcut "M") to adjust the volume and panorama of the mixer channels as well as automate fader and panorama movements. There are also a variety of plug-ins, submix and AUX buses, a four-band parametric equalizer, inserts for the integration of effects and VST instruments as well as many configuration and bouncing options available for individual channels as well as the stereo master output.

Each arranger track corresponds to a channel on the mixer.

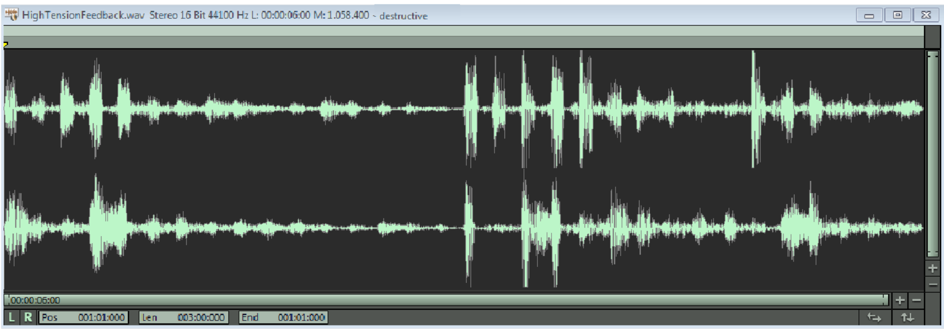
Detailed information about this is available in the chapter "**Mixer** (view page 203)".

Audio file editing

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

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

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



Audio file window: The audio file is opened in a separate window and displayed as a waveform. The function of the zoom, scroll, range and marker features does not change for the new window. In the title bar of the audio file window, you will see the name of the file, the bit resolution, the length of the sample and the resulting required memory.

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

Mouse modes: The following mouse modes are available for audio editing: Range Mode, Wave Draw Mode (view page 145), Volume Draw Mode, Scrub Mode, Zoom

Mode, Spectral Mode and Automation Draw Mode for realtime audio editing (view page 687).

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

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

> Detailed information about destructive editing mode and realtime editing mode can be found in the menu reference under „Object“ > „Wave Editing...“ (view page 686).

Effects

In Samplitude, effects can be used at various "levels", offline or in realtime, for objects, tracks in the mixer channel or as master effects.

Offline Effects

These effects can be used in audio files and on objects. Objects in a virtual project reference this audio file. Offline effects change the audio data and can be set through "Effects > Process effects offline" (view page 981).

> Detailed information about offline effects can be found in the "Effects – organization and workflow (view page 236)" chapter.

Real-time Effects

Unlike offline effects, real-time effects are not calculated into the wave files that the objects refer to. These effects are recalculated every time they are played and can be modified, varied and switched without changing your original audio.

> Detailed information about real-time effects can be found in the "Effects – organization and workflow (view page 236)" chapter.

Effect Categories

In general, the following effect categories are available in objects, tracks, and in the master:

- Dynamics
- Frequency / Filter
- Delay / Reverb
- Distortion
- Restoration
- Stereo / Phase
- Modulation / Special
- MAGIX Plug-ins
- VST FX (only available if the VST effects are installed)

Effects in Audio Objects (Object Effects)

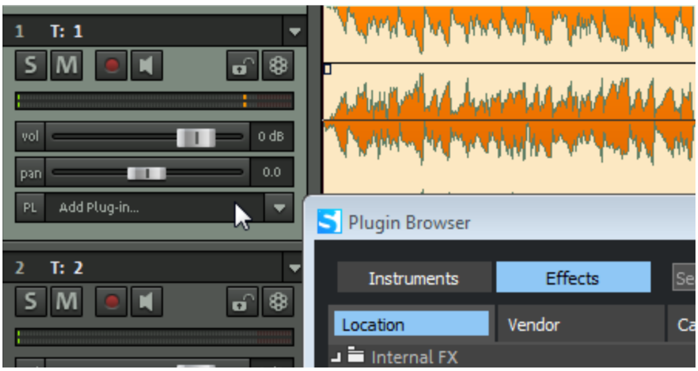
Object-based virtual effects are set in the Object Editor. You can open the editor by double-clicking on the object. The effect view is opened here as default.

Object effects are applied only to the selected object. All other objects in the VIP are not affected by these settings.

Effects in tracks (track effects)

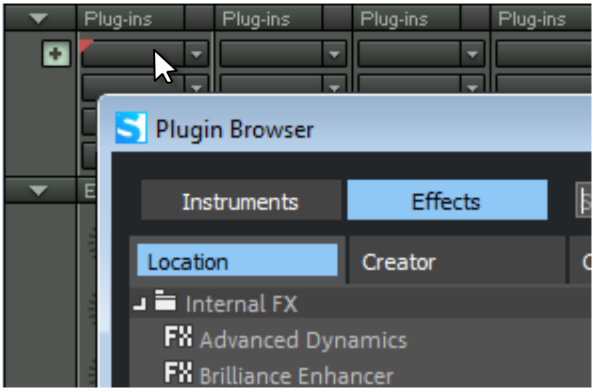
Track-related effects can be set either in the project window or in the mixer. These can be internal effects or VST effect plug-ins.

When you click on the plug-in slot in the track header of the track, the plug-in browser will open with the various effects.



Mixer Effects

To set effects in the mixer, open the mixer by pressing the "M" key. In the "Plug-ins" section, click the effect slot in the corresponding channel. This opens a plug-in browser for selecting effects. Use the "FX" button directly below the track's channel fader to open a dialog to change the series of effects within the track.



Effects in the Master

Master effects are applied to all of the tracks in a multitrack arrangement. They are basically applied right before the signal reaches the master output of your sound card.

The master effects can also be set in the mixer. The master area of the mixer is wider than normal tracks and can be identified by the red faders.

Here you will find the master plug-in slots for internal effects, MAGIX plug-ins and VST effects.

The **sequence of the master effects** can be set in the **FX routing dialog**, which can be opened with the FX button beneath the master faders.

MP3/AAC Preview Plug-in (view page 845): In the master plug-ins section under "Modulation/Special", there is a preview function available which can be used to prepare MP3s or AAC files for export. This makes it possible to hear in realtime what the generated file will sound like after encoding. You can change the export settings directly in the dialog window. During the mastering process, you have the option of taking the idiosyncrasies of the encoder into account and adjusting the quality relative to the requirements (e. g. "Mastered for iTunes").

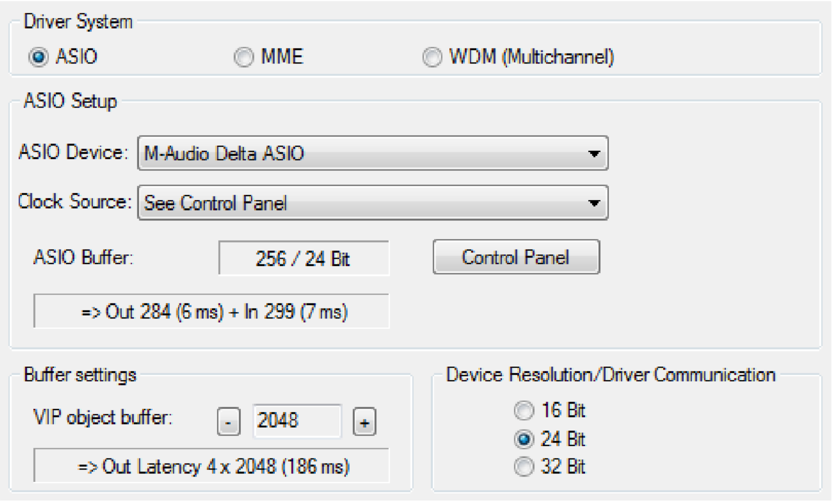
System settings

You can configure settings for your system and sound card in the system settings. Press the "Y" key on your keyboard or open the global system settings via "File" > "Program preferences" > "System/Options".

This allows you to access information about any connected playback and recording devices as well as MIDI, metronome, or program settings. You can also select the target folders for projects and VST plug-ins and the view options and coloring of screen elements. The most important dialogs are explained briefly in the following.

Audio Settings

Enter all essential settings for the driver system, buffers, driver communications, and monitoring here.



Driver system: A so-called "driver system" is used for the necessary communication between Samplitude and your sound card. In order to take full advantage of the program, we recommend that you use ASIO.

MME is the standard Windows multimedia driver system with the best compatibility. It also supports 16-bit playback. If you are recording 24/32-bit audio material, you can use MME/WDM. This driver system is suitable for multitrack recordings of up to 64 tracks that don't place severe requirements on the monitoring processes. For performance critical recordings, this provides greater security compared to ASIO drivers but with many sound cards, multitrack recordings will not be synchronous.

ASIO: Use a sound card model fitted with ASIO drivers if possible. This offers a number of decisive advantages over the MME/WDM driver system:

- Lower latency (input/output delay) in terms of the driver system. Resulting response times during real-time editing are clearly reduced. This makes it possible to use software monitoring for inputs and VST instruments.
- ASIO is intended for editing multitrack recordings featuring several sound cards that are all using the same ASIO drivers. The sound cards are synchronized by the ASIO driver.
- Advanced hardware monitoring options are also available by using ASIO direct monitoring.

ASIO settings

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

Buffer settings: The VIP object buffer indicates the buffer size for internal processing of object effects and economy tracks in hybrid mode. In every other monitoring mode, the buffer size also determines the editing of track effects. With a smaller VIP object buffer size, the amount of playback delay also decreases (latency). In these cases, the processor may become overloaded, which will result in dropouts during playback. Large buffer sizes actually increase stability, but they also increase the latency of the system. The field below displays the resulting latency in relation to the buffer settings.

Tip: The VIP object buffer size should normally be bigger than the ASIO buffer size and be set between 1024 and 8096 samples.

Device Resolution/Driver Communication: This option enables you to select the bit resolution for communication with audio devices. The preset value correlates with that of the sound card installed on your system.

The ASIO driver system specifies the bit resolution of the ASIO drivers. Samplitude always assumes the bit resolution set for the ASIO drivers. The settings options displayed for device resolution/driver communication only specifies the bit depth applied by dithering (view page 638). For 32-bit data transfer, dithering is 24-bit or 16-bit. The settings for 32-bit are recommended especially for Samplitude

- if no dithering is intended.
- if the input is supplied directly by a DSP card instead of from the converter unit of the sound card.

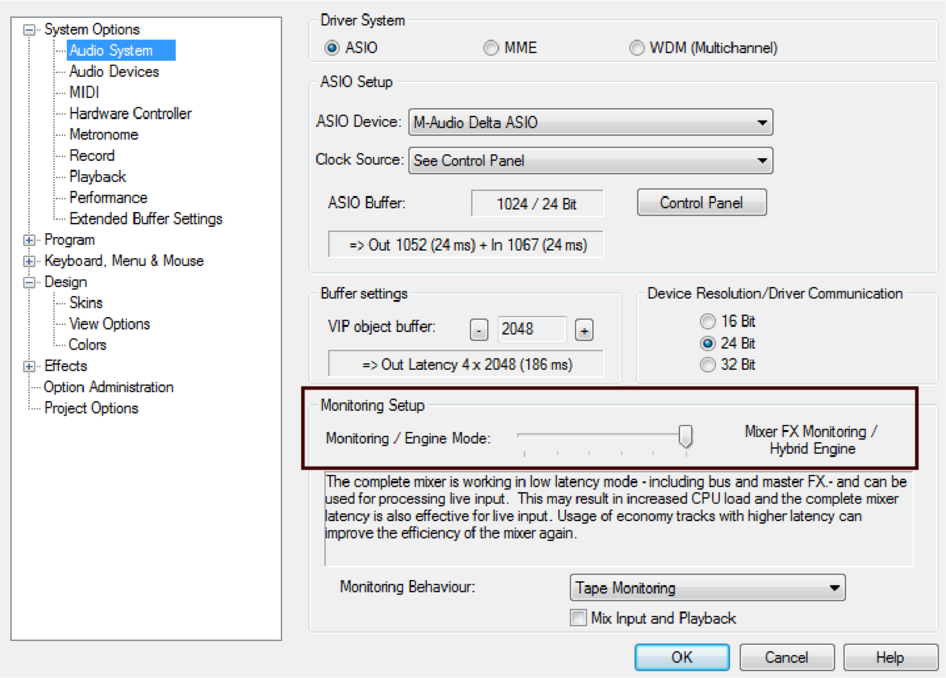
If MME driver system is selected, Samplitude opens the drivers with the bit depth set under Device resolution/Driver communication. If the output device cannot process the set bit resolution, the corresponding lower resolution will be produced and this will be transmitted to the MME driver.

To specify the plug-in buffer and the path to your VST plug-ins, switch to the dialog "System options (Y) > Effects > VST/ReWire (view page 615)".

Monitoring settings

Monitoring is essentially the process of making the input signals audible, e.g. so musicians who are playing on the recording can hear what they are doing. The routing of monitoring signals is an important and complex application that has various requirements.

The dialog can be found in the System Options (keyboard shortcut: Y) under "Audio Setup > Monitoring Setup".



For each selected monitoring setting a matrix display also opens which shows which buffer sizes are used and what effect the currently selected engine mode has on the latencies.

Overview of monitoring and engine modes

No Audio Monitoring (peak meter only): Displays the input level, but cannot be heard. This monitoring option is useful if you are using an external mixer for audio monitoring.

Used Buffers

	Economy			Hybrid	
	Peakmeter Monitoring	Hardware Monitoring	Software / Track FX Monitoring	Hardware Monitoring	Mixer FX Monitoring
Object	VIP	VIP	VIP	VIP	VIP
Track/VSTi	VIP	VIP	VIP	ASIO	ASIO *
Track input	-	HW	ASIO	HW	ASIO
VSTi input	-	ASIO	ASIO	ASIO	ASIO
Busses/ Master	VIP	VIP	VIP	ASIO	ASIO

HW = monitoring in sound card (no latency)
ASIO = ASIO buffer (low latency)
VIP = VIP Object buffer (higher latency)
* = with Economy Tracks VIP buffers will be used

With this matrix display of Peakmeter Monitoring, input signals on tracks or VST instruments cannot be heard in Samplitude. Already existing objects, (VSTi) tracks, as well as all busses and the stereo master output (Master) will still be played back through the monitoring by means of a VIP object buffer with high latency.

Hardware Monitoring: Audio monitoring through the sound card. This is the only possible monitoring type for MME drivers; For ASIO, many sound cards are able to process functions such as mute/solo, volume, and pan directly. For large ASIO or VIP buffers, minimal latency can be achieved with just a few samples. Effects cannot be applied to the input signal.

Used Buffers

	Economy			Hybrid	
	Peakmeter Monitoring	**Hardware Monitoring**	Software / Track FX Monitoring	Hardware Monitoring	Mixer FX Monitoring
Object	VIP	**VIP**	VIP	VIP	VIP
Track/VSTi	VIP	**VIP**	VIP	ASIO	ASIO *
Track input	-	**HW**	ASIO	HW	ASIO
VSTi input	-	**ASIO**	ASIO	ASIO	ASIO
Busses/ Master	VIP	**VIP**	VIP	ASIO	ASIO

HW = monitoring in sound card (no latency)
ASIO = ASIO buffer (low latency)
VIP = VIP Object buffer (higher latency)
* = with Economy Tracks VIP buffers will be used

With this matrix display for hardware monitoring, the input signals on tracks are played back directly through the sound card and not in Samplitude which results in almost latency-free monitoring. VSTi input signals are played back within Samplitude with low latency by means of an ASIO buffer. Already existing objects, (VSTi) tracks as well as all busses and the stereo master output (Master) can be monitored by means of a VIP object buffer with high latency.

Software Monitoring/Economy Engine: Audio monitoring that takes into account the recording track levels and the playing of software instruments. This monitoring option is only available when using ASIO drivers. No effects are applied to the input signals which means that latency and CPU load remain the same even in complex projects.

Used Buffers

	Economy			Hybrid	
	Peakmeter Monitoring	Hardware Monitoring	**Software / Track FX Monitoring**	Hardware Monitoring	Mixer FX Monitoring
Object	VIP	VIP	**VIP**	VIP	VIP
Track/VSTi	VIP	VIP	**VIP**	ASIO	ASIO *
Track input	-	HW	**ASIO**	HW	ASIO
VSTi input	-	ASIO	**ASIO**	ASIO	ASIO
Busses/ Master	VIP	VIP	**VIP**	ASIO	ASIO

HW = monitoring in sound card (no latency)
ASIO = ASIO buffer (low latency)
VIP = VIP Object buffer (higher latency)
* = with Economy Tracks VIP buffers will be used

With this matrix display of Peakmeter Monitoring, input signals on tracks or VST instruments can be played back in Samplitude with low latency by means of an ASIO buffer. Already existing objects, (VSTi) tracks as well as all busses and the stereo master output (Master) can be monitored by means of a VIP object buffer with high latency.

Track FX Monitoring: Audio monitoring including the track effects of the recording track. This monitoring option is only available when using ASIO drivers. Effects applied directly to the recording track are included in the monitoring signal, but bus and master effects are not.

Used Buffers

	Economy			Hybrid	
	Peakmeter Monitoring	Hardware Monitoring	Software / **Track FX** Monitoring	Hardware Monitoring	Mixer FX Monitoring
Object	VIP	VIP	**VIP**	VIP	VIP
Track/VSTi	VIP	VIP	**VIP**	ASIO	ASIO *
Track input	-	HW	**ASIO**	HW	ASIO
VSTi input	-	ASIO	**ASIO**	ASIO	ASIO
Busses/ Master	VIP	VIP	**VIP**	ASIO	ASIO

HW = monitoring in sound card (no latency)
ASIO = ASIO buffer (low latency)
VIP = VIP Object buffer (higher latency)
* = with Economy Tracks VIP buffers will be used

With this matrix display of Track FX Monitoring, input signals on tracks or VST instruments can be played back in Samplitude with low latency by means of an ASIO buffer. Already existing objects, (VSTi) tracks as well as all buses and the stereo master output (Master) can be monitored by means of a VIP object buffer with high latency.

Hardware Monitoring/Hybrid Engine: In this case, the complete mixer works in low latency mode. This keeps playback latency to a minimum. The input signals are monitored via the sound card in use.

Used Buffers

	Economy			Hybrid	
	Peakmeter Monitoring	Hardware Monitoring	Software / Track FX Monitoring	**Hardware** Monitoring	Mixer FX Monitoring
Object	VIP	VIP	VIP	**VIP**	VIP
Track/VSTi	VIP	VIP	VIP	**ASIO**	ASIO*
Track input	-	HW	ASIO	**HW**	ASIO
VSTi input	-	ASIO	ASIO	**ASIO**	ASIO
Busses/ Master	VIP	VIP	VIP	**ASIO**	ASIO

HW = monitoring in sound card (no latency)
ASIO = ASIO buffer (low latency)
VIP = VIP Object buffer (higher latency)
* = with Economy Tracks VIP buffers will be used

With this matrix display for Hardware Monitoring through the Hybrid Engine, the input signals on tracks are played back directly through the sound card and not in Samplitude which results in almost latency-free monitoring. VSTi input signals are played back within Samplitude with low latency by means of an ASIO buffer. (VSTi) tracks as well as all busdes and the stereo master output (Master) can be monitored by means of an ASIO buffer with low latency. Already existing objects will still be played back through the monitoring by means of a VIP object buffer with high latency.

Mixer FX monitoring/Hybrid Engine: The Hybrid Audio Engine enables audio monitoring throughout the entire mixer and, in doing so, also calculates playback tracks in the mixer with short latency. This way you can also mix data from the hard drive with the lowest possible playback delay. We recommend this mode for input signals as well as for live mixing with hardware controllers because access to all bus and master effects is guaranteed.

Used Buffers

	Economy			Hybrid	
	Peakmeter Monitoring	Hardware Monitoring	Software / Track FX Monitoring	Hardware Monitoring	**Mixer FX** Monitoring
Object	VIP	VIP	VIP	VIP	**VIP**
Track/VSTi	VIP	VIP	VIP	ASIO	**ASIO** *
Track input	-	HW	ASIO	HW	**ASIO**
VSTi input	-	ASIO	ASIO	ASIO	**ASIO**
Busses/ Master	VIP	VIP	VIP	ASIO	**ASIO**

HW = monitoring in sound card (no latency)
ASIO = ASIO buffer (low latency)
VIP = VIP Object buffer (higher latency)
* = with Economy Tracks VIP buffers will be used

With this matrix display of Track FX Monitoring through the Hybrid Engine, input signals on tracks or VST instruments can be played back in Samplitude with low latency by means of an ASIO buffer. (VSTi) tracks as well as all busses and the stereo master output (Master) can be monitored by means of an ASIO buffer with low latency. Already existing objects will still be played back through the monitoring by means of a VIP object buffer with high latency.

The following applies to all monitoring modes: The set track effects on the recording track are not included in the recording which means that the track actually recorded does not have any of the effects that have been applied to the track. Depending on the monitoring mode used during the recording process – and during playback of the recording, of course – you will hear the recorded signal with the effects applied to it. For example, it is not possible to record the vocal signal with the Samplitude reverb VariVerb II directly. However, you can save the recorded track afterwards along with all track effects using the "Track Bouncing (view page 582)" function.

Tip: In most cases we recommend using the "Mixer FX Monitoring/Hybrid Engine". This monitoring mode enables audio monitoring throughout the entire mixer and also calculates playback tracks in the mixer with short latency. This also gives you access to all the bus and master effects.

If the CPU load becomes too high when using "Mixer FX Monitoring/Hybrid Engine", switch to the Monitoring/Engine Mode that best meets your requirements and can be handled by your system.

For example, if you want to do without effect monitoring of input signals, "Software Monitoring/Economy Engine" might be the right mode for you.

If you want to include the monitoring of track effects of the input signals and to do without the bus and master effects in the monitoring path, we recommend the "Track FX Monitoring" mode.

If you prefer to monitor to the track input signals through the sound card, you can choose either "Hardware Monitoring" or "Hardware Monitoring/Hybrid Engine" depending on the performance capability of your system.

Note: If you use the "TotalMix" software from RME in combination with a hardware monitoring mode, set the panning law in "TotalMix" to -6 dB. This will ensure that the recording level in Samplitude corresponds to the monitoring level in "TotalMix".

Hybrid Audio Engine

In general, "Hybrid" refers to a system in which two separate technologies are combined with one another. Samplitude's Hybrid Audio Engine is a special combination of a Low Latency Engine and the classic Samplitude Playback Engine with higher latency. The Low Latency Engine calculates live input signals and the output of the playback engine. This provides short reaction times for calculating track effects and also provides monitoring with low latency. The classic playback engine saves resources and enables the integration of high-performance effects.

The Hybrid Engine provides a combination of Low Latency Engine and a classic engine for calculating the track and effects. For example, processor-intensive VST instruments can be played on the so-called "Economy" tracks (see below), while only the VST instrument that is being recorded is calculated by the Low Latency Engine.

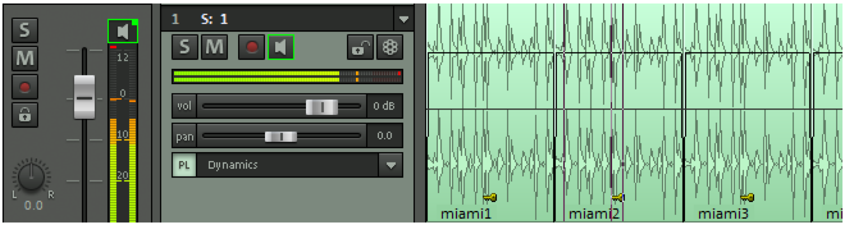
Go to System Options (shortcut "Y") ->"Audio System" -> "Driver System" and select "ASIO". "Monitoring Setup" features the mode "Mixer FX monitoring/Hybrid Engine".

Mixer FX monitoring provides audio monitoring including all insert and AUX effects and also calculates the playback tracks with the least amount of latency. This means that the entire mixer can operate in "Low latency" mode, including the bus and master effects, and it can also be used for editing input signals.

Economy Track

If you have selected the ASIO Hybrid Engine as the driver system, you can take individual tracks from the Low Latency Engine and calculate the track effects with a larger VIP buffer size for VIP objects during playback. This prevents the system processor from being overloaded but can lead to an increased delay time during playback.

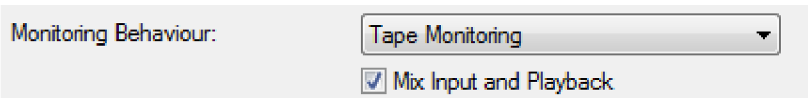
To switch a track to "Economy", select the option "Track -> Track type -> Economy Track".



The "Volume" button for the economy track is framed in green.

> **Note**: The Hybrid Engine compensates for latency of the track effects in economy tracks so that the total latency of the mixer does not increase for other tracks.

Monitoring switch behavior

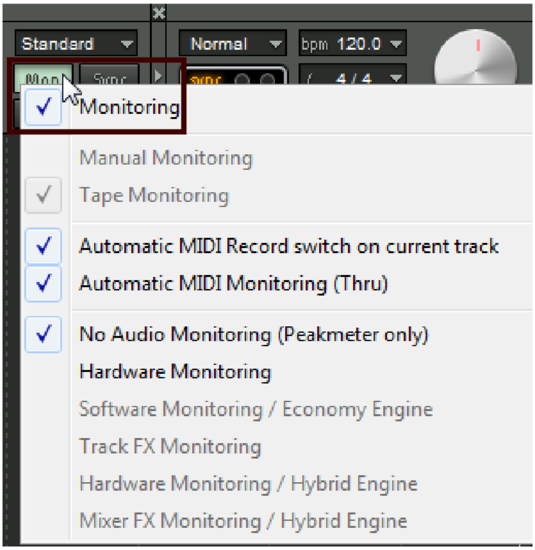


Tape Monitoring is the preset monitoring behavior and the monitoring behavior familiar from earlier Samplitude versions which works like a tape device: In stop status the input signal is played back; in play status the track content is played back. During a recording, the input signal is played back, except for punch-in recordings; here, outside the punch marker, the track content is output and inside the punch marker, the input signal is output.

Manual Monitoring: For each track you can manually specify whether the monitoring is active for the track using the loudspeaker button in the track header, track editor or mixer. This switch behavior is only available when the ASIO driver system is used.

Mix Input and Playback: If you set the check here, the input signal can also be heard when playback is running if track monitoring is active.

The monitoring settings and switch behavior can be set in "System Options > Audio System" and can also be accessed directly by right-clicking on the "**Mon**" button. This displays two more MIDI recording options:

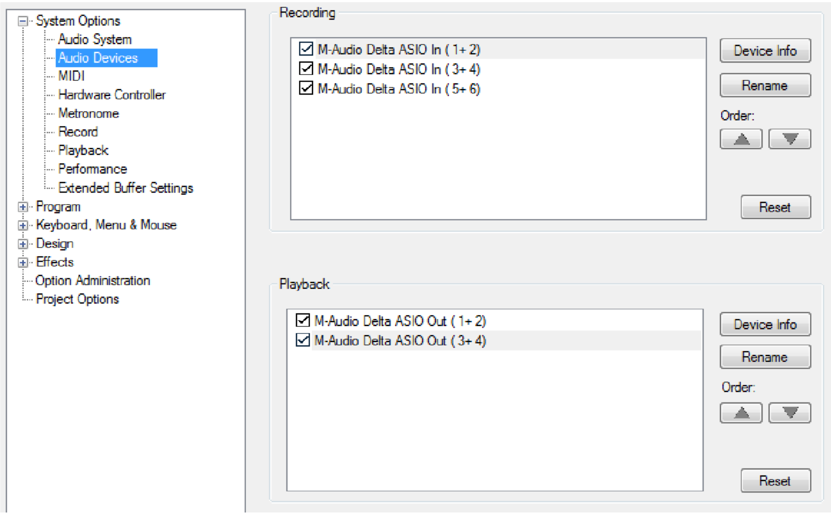


Automatic MIDI record switch on current track: This option ensures that MIDI tracks are always ready for recording as soon as they are selected. This is indicated by the pink recording buttons.

Automatic MIDI monitoring (thru): If this option is selected, every MIDI track that you activate for recording will be switched on automatically, e.g. if you are playing a MIDI keyboard that is being recorded on this track, you will always immediately hear the output signal of the software instrument for this track.

Audio Devices

This dialog is where you select the inputs and outputs for the installed sound cards that will be used in Samplitude.

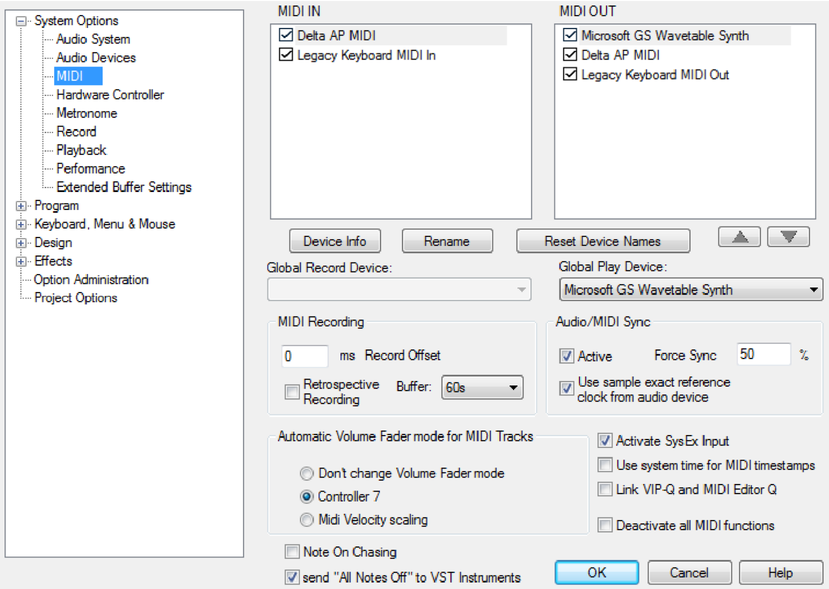


You can activate or deactivate a device by adding or removing a check mark in the box next to it. You can set the sequence with the arrow keys. The "Reset" button activates all of the devices displayed. With ASIO, only the first 4 stereo channels are activated in the default setting. However, if you hold down the Shift key, all of the inputs and outputs will be activated. Clicking on the "Info" button opens the control panel for your sound card. With the "Rename" button you can customize each device name individually.

MIDI Settings

Tip: For smooth playback of VST instruments, please always use the monitoring settings "Track FX monitoring", "Hardware monitoring/Hybrid Engine", or "Mixer FX monitoring/Hybrid Engine".

Global MIDI Devices



Open the System Options – MIDI through "File" > "Program Preferences > MIDI Options" and set the standard input and standard output. The set devices are used to play back MIDI files directly, and they are also used as the standard settings for new tracks. You can rename the devices at any time.

Record offset

Enter a recording offset value in milliseconds to determine the time difference which passes between a MIDI command and placement in the arrangement.

Samplitude places played notes in the MIDI object on the right according to the set time, i.e. MIDI notes are delayed and displayed later in the timeline.

Retrospective MIDI recording

Activate the "Retrospective recording" option here for the same function available under "Play/Rec -> Retrospective MIDI recording (view page 748)". Samplitude creates a MIDI object on the MIDI track that can be adjusted in terms of buffer length.

Audio/MIDI synchronization

Force sync specifies how precisely Samplitude synchronizes MIDI tracks with audio tracks. On faster systems, this setting should be at 100% in order to achieve the closest sync between the MIDI and the audio. If your system experiences difficulties synchronizing the MIDI tracks with the audio tracks, select a lower sync reciprocation value. Normally you will also want to use sample-exact reference times from your audio device (e.g. sound card) for audio/MIDI synchronization.

Automatic volume fader mode for MIDI tracks

The behavior of the MIDI track's volume fader is preset to controller 7 (MIDI volume).

Alternatively, you can also select that "Volume fader" mode should not change for MIDI tracks or that the volume fader for MIDI tracks should match the MIDI velocity scaling. New MIDI tracks will assume the currently set Volume fader mode (view page 206).

Activate SysEx input

Selecting this option causes Samplitude to receive SysEx data from external devices.

Use system time for MIDI timestamps

If this option is activated, the MIDI device driver's timestamp will be ignored. This is helpful if drivers provide a timestamp that is not synchronized with the audio or is completely incorrect.

This function detects invalid driver timestamps and reverts back to the system time automatically. This helps fix MIDI recording problems with MIDI devices.

Link VIP Q and MIDI editor Q

This option links the grid or quantization settings between the VIP and MIDI editor, i.e. changing the grid/quantization settings in the VIP will automatically be assumed for the grid/quantization settings in the MIDI editor and vice versa.

Deactivate all MIDI functions

You can shut off all the MIDI functions in MIDI or VIP projects. If you do this, the MIDI functions in the arranger, track editor and track settings dialog will no longer be available.

Note On Chasing

"Note On Chasing" ensures that held MIDI notes are played back even if they are positioned before the current playback start position.

Send "All Notes Off" to VST instruments

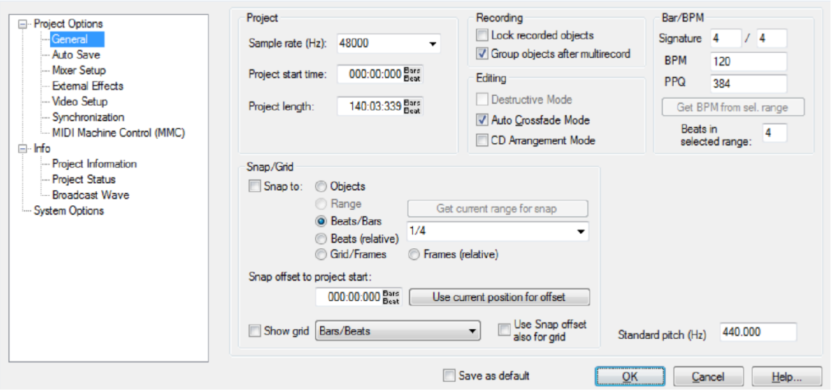
As an alternative to the "All Notes Off" mechanism, an individual "Note Off" mechanism is available for VST instruments, since "All Notes Off" is ignored by some

VST instruments. You can completely shut off "All Notes Off for VSTis in the MIDI settings.

Project Options - General

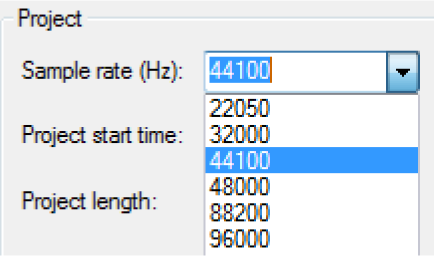
Shortcut: I

Here you can see the most important information about the current project at a glance. This includes Bar/BPM, Recording, Editing, and Snap/Grid settings.

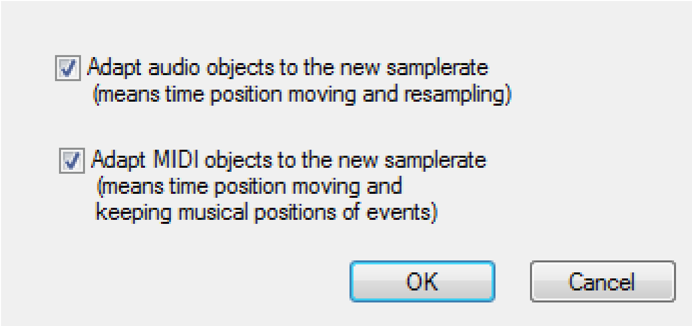


Project Options - General

Sample rate (Hz): This indicates the project's sample rate. You can also change the sample rate for the project



and adjust audio and MIDI objects to the modified sample rate.



Project start time: Specify the project's starting time here.

Project length: The project length is displayed here in bars and beats.

General project options – recording/editing

Here you'll find various presets for recording and editing:

Lock recorded objects: Protects against unintentional moving of recorded objects.

Group objects after multirecord: Objects that belong together will be grouped in a multitrack recording.

Destructive editing mode (only for audio files): Detailed information about destructive offline audio editing can be found in the chapter "Basic Terms > Audio Editing in Samplitude" (view page 66).

Auto Crossfade Mode: Use this function to activate a mode which adds a crossfade to all newly recorded objects, objects created from cuts or audio files that you imported to your project. Every object can be assigned a standard fade-in which can be edited in the "Object editor fade" menu under "Get/set global crossfade". If two objects overlap in this mode, a realtime crossfade will occur at the intersection.

CD Arrangement Mode: If this menu item is activated, Samplitude arranges newly added objects to insert a "Red Book Standard" pause between the objects. "Red Book" is a standard for audio CDs that sets the technical specifications for all CD and CD ROM formats.

Project options - Bar/BPM

Bar/BPM/PPQ: Enter the time signature (counter/signature), the tempo in beats per minute (BPM), and the timer resolution in peaks per quarter (PPQ/clicks per quarter note) here.

Get BPM from selected range: If you enter the number of beats into the field beside "Selected range has beats:", Samplitude will calculate the BPM based on the selected range when the button is pressed.

General Project Options – Snap / Grid

Snap to: Switches the global snap on/off.

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

Range: Activates the range snap and enables the current range to be used as the basis for snapping.

Beats/Bars: Activates a grid with bars as the basis for snapping.

Beats (relative): Also this option activates a grid which uses beat intervals as its basis. For example, here you can move a selected object while maintaining its relative distance to the closest marker according to the set bar grid.

Grid/Frames: This option activates a frame-based grid.

Frames (relative): This option also activates a frame-based grid where the object maintains its relative distance to the closest grid marker when moved.

Snap offset to project start: Sets the snap offset relative to the beginning of the project. "Use current position for offset" specifies the current position of the playback marker as the grid's zero position.

Show grid: If a check is placed here, the grid will be displayed for the project according to the snap unit set in the selection box beside it.

Use snap offset also for grid: The snap offset is used as a reference size for the grid.

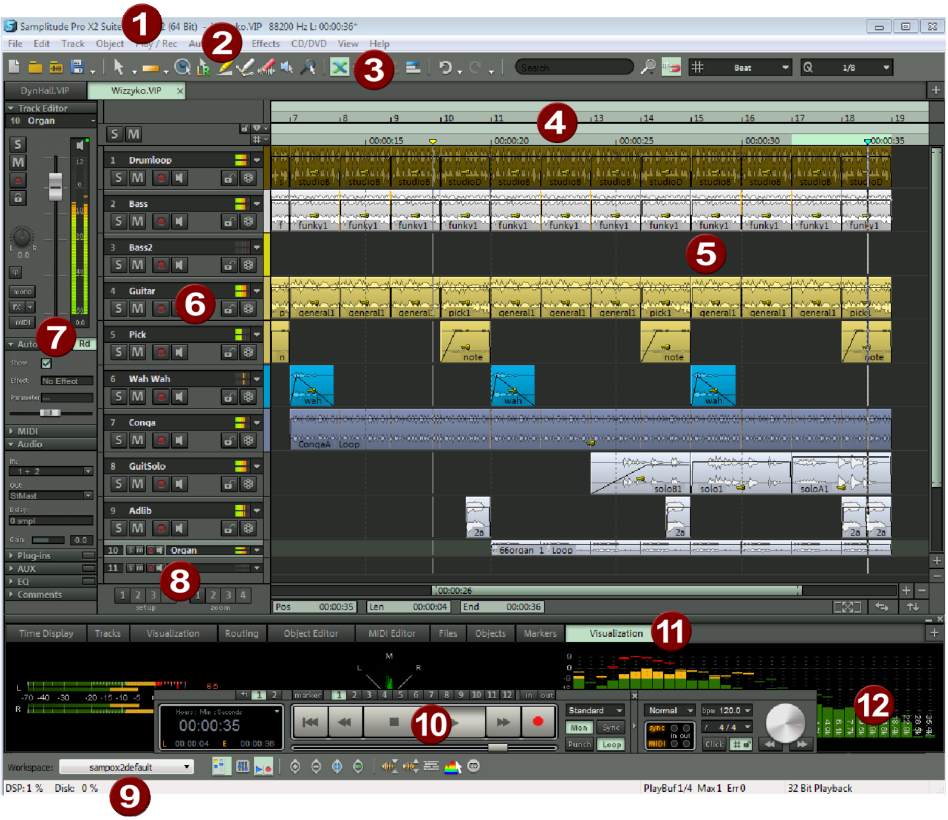
General Project Options - Standard pitch for tuner

This field indicates a standard pitch of A at 440 Hz. You can change this setting if you want to use an alternative tuning for the Samplitude internal tuner (view page 1029).

Screen Elements

Program interface – Overview

VIP Window:



1 **Title bar**: The title bar is located at the top of the window. The program name can be seen here along with the name, sample rate and length of the current project.

2 **Menu bar**: The menus are located directly below the title bar. A keyboard shortcut (view page 538) can be assigned to each menu item.

> Detailed information about how to use menu items and keyboard shortcuts is provided in the menu reference, help file, or in the "manual.pdf" documentation under "File -> Program Preferences -> Edit Keyboard Shortcuts and Menus (view page 626)".

3 **Toolbars**: Toolbars consist of buttons that execute specific commands or specify states. They are ordered above and below the arranger in groups. You can move a toolbar group by clicking the separator on the left and dragging with the mouse.

Right-clicking an icon opens a **context menu**. Here you can hide the selected bar or display it with large icons. "**Edit Toolbar**" opens a dialog to individually remove current buttons or add other available buttons for the selected bar.

4 **Grid/marker bars**: The grid/marker bars are located above the first track in the project. In the upper half (if two grid bars are displayed between both grid bars), you'll find the marker bar, where you can position the markers and playback cursors. The grid bars display the project time in relation to the selected measurement unit. You can also define various ranges.

5 **Arranger**: Displays the project tracks together with the audio and MIDI objects. There are plenty of options for you to scroll (view page 128) and zoom (view page 121) through the visible section of the arranger.

6 **Track header:** The track header includes important control elements such as mute and solo function and recording next to the track names. At higher vertical zoom levels, some elements are faded out. More information about the individual controls of the track header can be found under "Track header" (view page 97).

7 **Track Editor**. The Track Editor is arranged far left next to the track headers. This lets you access all important settings for the selected track. More information about the individual controls can be found under "Track Editor" (view page 92).

8 **Position & Parameter Fields :** The fields "Pos" for position/range start, "Len" for range length and "End" for range end can be configured using the right mouse button. In the context menu that opens you can also display additional parameters in up to 5 fields, e.g. the current mouse position or the last edited mixer setting.

9 **Status display**: The status display appears at the bottom border of the VIP window. Here you'll find information about CPU load, latency, buffer, and current operations like loading, saving, effect calculation, etc. You can also open the status display by going to "View" > "Toolbars" > "Status Display".

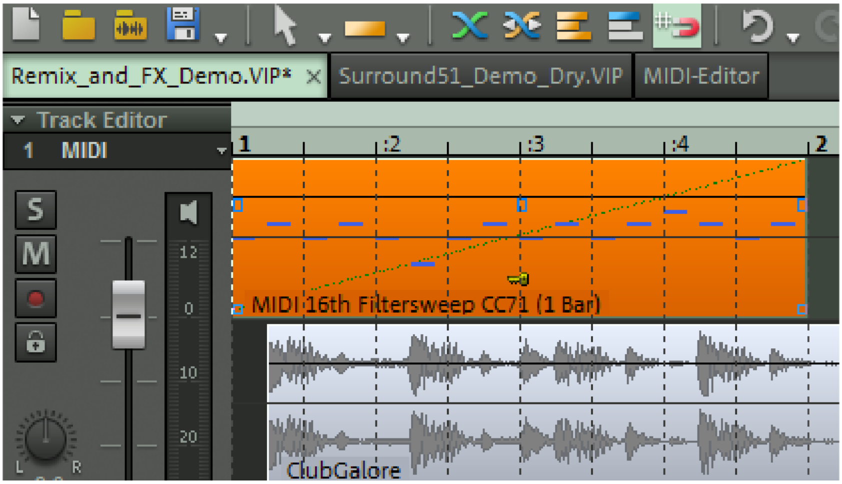
10 **Transport console** (view page 90)

11 **Manager/Docker** (view page 90)

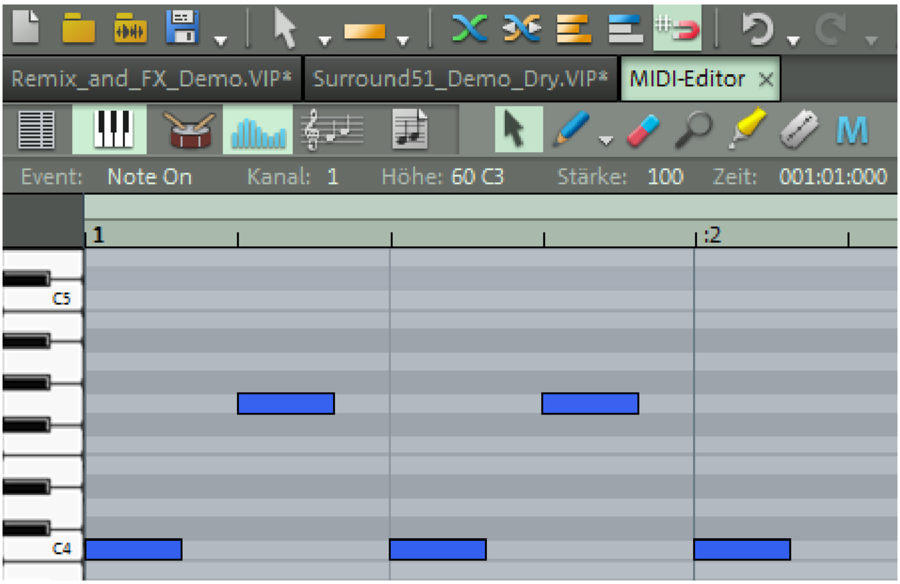
12 **Visualization** (view page 1011)

Arranger Tabs

All opened projects will be displayed as tabs. You can quickly switch between projects by clicking on tabs.



Each tab window from the docker (view page 90) can also be displayed as an Arranger tab. Simply drag the desired video from the Docker into the Arranger tab area. This way you can, for example, drag the MIDI Editor tab from the Docker into the Arranger tab to quickly switch between the Arranger view and the MIDI Editor in the main screen.



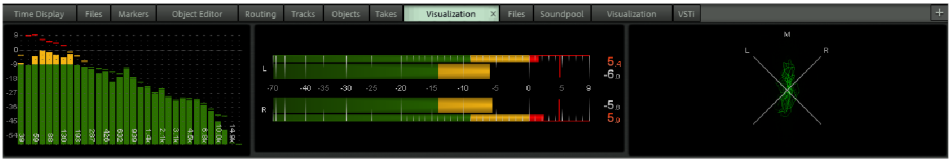
Or you can drag a project from the Arranger tab to a second monitor to view two projects at once.

Using the "+" key on the right of the Arranger tab area you can additionally create new VIP projects.

Manager/Docker

Menu: "View" > "Manager / Docker"

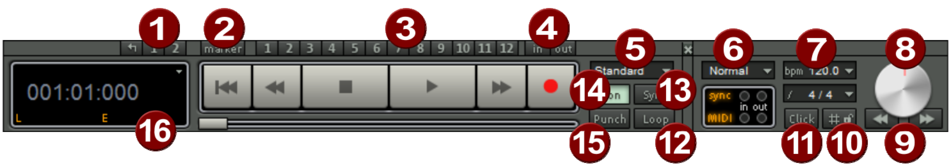
The docker is where you can customize the display windows you need for your workflow. In addition to the manager windows (view page 174), now the visualization, time display, transport console (view page 90), object editor (view page 146), MIDI editor (view page 329), Monitoring section and audio quantization (view page 694) are opened as a tab window in the docker.



More information about docking can be found in the "Working in the Project Window > Docking" chapter.

Transport console

Shortcut:	Ctrl + Shift + T



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

1. **Ranges:** Use the buttons "1" and "2" to save ranges. The arrow symbol can be used to open previously saved ranges.
2. **Marker:** Opens the Marker manager (view page 184).

3 **Marker 1-12:** The current playback position can be saved to one of the 12 marker buttons by clicking on it. If one position has been saved, the marker will appear colored. Another click on the same marker moves the playback position to the marker. You can delete a marker by right-clicking on it.

4 **Punch In/Out:** Sets the in/out point for a punch recording. Additional Punch In/Out markers can be set by holding down the Alt key.

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

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

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

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

6 **MIDI record modes**: Select from the following MIDI record modes: normal, overdub, multi-overdub, and replace (view page 51).

7 **Tempo/Beat section:** You can change the playback speed and the tempo of the arrangement here. Adjust (view page 394) all objects to the current speed value by moving or stretching them.

8 **Scrub Control:** Using the Scrub Control wheel you can variably set playback speed. This is helpful when searching for specific audio passages. Right-clicking opens the scrubbing and Varispeed options (view page 143).

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

10 **Grid**: You can activate the beat grid and have it displayed (view page 99) here.

11 **Click:** Use to switch on the metronome click. A right-click opens the metronome options window (view page 44) to configure the pre-counter and click volume.

12 **Loop**: Switch to Loop mode here (view page 49).

13 **Sync**: This button opens a dialog with the Synchronization options (view page 450).

14 **Mon:** Activates monitoring. All tracks with activated recording display the incoming signal on the peak meter. Right-clicking on the "Mon" button opens a selection of monitoring modes. You can find out more by reading "Monitoring settings" (view page 72).

15 **Punch:** Switches Samplitude to punch marker mode.

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

Track Editor

This function has been revised in Samplitude. For the latest information, see the PDF document **Samplitude Pro X7 New Features** in the program folder.

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

You can open the track editor using the corresponding button in Position bar 2 or through „View“ > „Track Editor“.

Keyboard shortcut: Ctrl + Alt + Shift + E

1. **Track number and track name:** Double-clicking on the track names allows them to be edited. Right-clicking the track name opens the "Track Options" dialog.
2. **The "S" button** mutes all tracks except the selected one (track solo function).

> Detailed information on the global solo modes can be found in "Global buttons -> Global solo modes (view page 224)".

3. **The "M" button** mutes the active track (track mute function).
4. **The "Rec" button** activates the track for the recording.
5. **The "Lock" button** protects objects in the track and prevents them from being unintentionally moved or deleted.
6. **The "Panorama" knob** controls the position of the channel within the mix. Right-clicking on this opens the Stereo panorama dialog (view page 229) where you can make adjustments to the stereo image, stereo width and the phasing.

7 **The "Phase reverse" button** reverses the signal phase 180 degrees. Right-clicking on this button also opens the Stereo panorama dialog (view page 229) where you can adjust further settings such as the stereo image, stereo width and the phase length of the track.

8 **The "Mono" button** switches the track to mono processing from the input up to the pan controller. In particular, all track effects preceding the pan controller operate in mono, which saves considerable CPU resources. However, the submix and AUX return buses always remain stereo. If stereo objects are located in mono tracks, the mono share (L+R) is generally played.

Here as well, right-clicking on the button opens the Stereo panorama dialog (view page 229) where you can make adjustments to the stereo image, stereo width and the phasing.

For standard routing, only the post DX/VST plug-ins and post AUX sends are arranged after the panorama controller and are therefore calculated as a stereo signal. The routing position of the pan controller, however, can be freely adjusted in the "FX routing" dialog.

9 **The "FX" button** opens the dialog for specifying the effect sequences and adding VST plug-ins; right-clicking this button also provides access to the track effects settings for the active track. These can be copied, inserted, reset, saved, or loaded.

Save your personal track effect settings in the program directory in "FX presets -> Track FX". You can also create new subfolders here. We've already included a selection of useful presets, e. g. for "Mid-side processing". The track effect settings of a VST instrument can also be saved (including parameters and all subsequent track effects), and transferred to other tracks.

10 **The "MIDI" button** switches the track to MIDI recording and opens the MIDI section (view page 95) of the track editor.

11 **Volume entry field and volume control**.

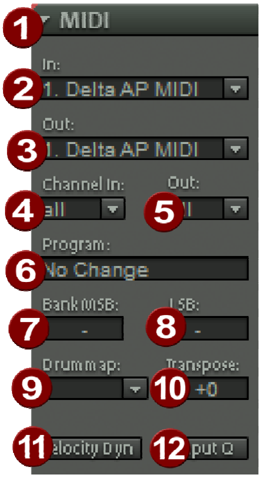
12 **The loudspeaker symbol** switches on monitoring (view page 79), i. e. the playback of incoming signals when the track record button has been activated. "MIDI thru" will be active here for MIDI tracks.

13 **Level control display:** Both LED rows display the input and output signal for the track.

14 **The "Automation" button** activates automation (view page 428) for the track.

15 **Automation parameter selection field and automation control**: Select automation parameters and adjust the values with the corresponding controllers.

MIDI



1. **Arrow**: Opens and closes the respective dialog box.
2. **In**: Opens the MIDI input menu.
3. **Out**: Opens the MIDI output menu.
4. **Channel In**: Set the MIDI input channel here.
5. **Channel Out**: Set the MIDI output channel here.
6. **Program**: This slot is used for choosing the MIDI instrument. The first click activates this field and the second mouse click opens the program selection. When the menu is open, use the arrow and page up/down keys to make your selection.
7. **Bank MSB**: Here you can set the MSB (Most Significant Byte) for the "Bank Select" MIDI message that controls your external instrument.
8. **LSB**: Here you can set the LSB (Least Significant Byte) for the "Bank Select" MIDI message that controls your external instrument. The bank number is MSB value* 128 + LSB value.
9. **Drum map:** Here you can select a drum map (view page 356) for allocating MIDI notes to the device-specific sound. If a drum map is selected, the transpose function will not be available because the pitch is firmly assigned by the drum map.
10. **Transpose**: You can transpose the notes up or down here. This function works in realtime meaning that the MIDI files in the MIDI objects will not be changed.

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

Detailed information about this **MIDI effect**, which processes the MIDI velocity dynamics, can be found in "MIDI editors -> MIDI functions -> Velocity dynamics (view page 338)".

12 **Input Q:** In this case, the MIDI quantize settings (view page 342) are used to quantize every recording offline. The original position can be restored by going to "Object -> Quantize -> Reset Quantization".

Audio

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

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

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

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

5 **Plug-ins**: Access track effect inserts, MAGIX plug-ins and VST effects here. Clicking on the empty insert slot opens the plug-in browser (view page 240), where you can load a plug-in to the slot. Click on an occupied slot to activate/deactivate the plug-in. Right-clicking on the slot opens the interface of the plug-in. The arrow next to the respective insert slot opens a menu with various functions. You can ope the plug-in browser again to switch out the plug-in or remove it ("No effect"). The "Plug-ins" button activates or deactivates all effects in the channel. A visual indicator (*) is displayed for plug-ins that were previously active and are active the next time "Plug-ins" button is pressed.

6 **AUX**: Here you can specify the input of the AUX sends (view page 225) or right-click to switch to the output assignments (More information can be found in the section "Assign track to multiple outputs" on page 212) or sidechain sends (view page 228) display.

7 **EQ:** This contains the parametric EQ for the track. Right-click to open an advanced input window with six frequency bands.

8 **Comments**: Section where you can quickly add notes.

Track Header

The higher the vertical zoom level, the fewer options the track header will display. In order to see all options, you need to zoom (view page 121) into the display.

1. **Track number and name:** Clicking the track name or the track number selects the track. Repeatedly clicking the track number causes the track's display height to switch between flat and high. Double-clicking on the track names lets you edit them. Right-clicking the track name opens the "Track Options" dialog. Click the arrow to select additional track-relevant functions from the context menu.
2. **The "S" button** (view page 224) mutes all tracks except the selected one (solo).
3. **The "M" button** mutes the active track (mute).
4. **Record**: This button (view page 41) activates the track for recording.
5. **The loudspeaker symbol** activates monitoring (view page 72).
6. **The lock button** prevents the objects in the track from being accidentally moved or deleted.
7. **Revolver tracks**: Revolver tracks (view page 142) can be used to compile objects differently for each individual track.
8. **Peak Meter:** Both LED rows display the input and output signal for the track.
9. **Vol:** This button switches on the volume automation curve (view page 428). This way, level adjustments in the track can be controlled via an automation curve and drawn by moving the channel fader.
10. **Volume** controller
11. **Pan**: Switch on panorama automation with this button.
12. **Panorama** control (view page 229).
13. **PL:** The plug-in selector (view page 236) assigns insert effects or VST plug-ins to a track.
14. **Color selection**: If you click on the right edge of a track header, a selection menu will appear for specifying the color (view page 113) of the track and its objects.
15. **Automation tracks:** Use the small triangle to fade in or out the automation tracks (view page 434).

Toolbars

Toolbars provide quick access to important functions in Samplitude.

Upper toolbar

1 New Virtual Project

2 Load VIP

3 Load audio file

4 Export/Save

5 List Field Mouse Mode

6 List Field Object Mode (view page 114)

7 Auto Crossfade On/Off

8 Crossfade Editor (view page 660)

9 Group

10 Ungroup

11 Grid On/Off and Grid Toolbar (see below)

12 Undo

13 Redo

14 metronome

15 Search for Command or Help Topic

Grid and Quantization menu

You'll find the grid and quantization menu in the toolbar. The grid menu lets you change important grid settings without always having to open project options (view page 84). Click the magnet button to activate the grid. The button beside it opens the grid menu with the following grid settings:

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

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

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

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

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

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

Here you must also specify the snap length (for beat/bar snapping only). The values range from "every 4 bars", "every 2 bars", "full bar", "beat" to "Snap to Quantization". The "Beat" setting means that signatures based on quarter notes serve as a quarter snap unit and time signatures based on eighth notes serve as an eighth snap unit. The snap therefore follows the time signature and observes any time signature changes.

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

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

Grid and snap settings: This option brings you to the **Project options - General dialog** (view page 84), where you can change additional grid and snap settings.

The field marked with the "Q" is where you open the quantization menu. Here you can specify the quantization value for MIDI (view page 342) and audio quantization (view page 694). The "Snap to Quantization" option sets the snap interval accordingly. This links the audio and MIDI quantization (view page 83). Right-clicking on the field opens the MIDI quantization settings.

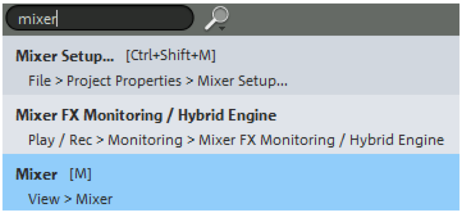
Note on "Input Q": In the MIDI section of the track editor beside the button for velocity dynamics you will find the "Input Q" button. If this function is active, recorded MIDI notes will be quantized immediately according to the current settings. The original position can be restored by going to "Object"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

Details about audio quantization are provided in the menu reference "Object > Quantize > Audio quantization wizard (view page 694)".

Searching for menu commands and help topics

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

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



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

Shortcut:	Ctrl + F

Lower toolbar

1 The Workspace (view page 118)

2 Track Editor

3 Open Mixer

4 Display transport console

5 Playback marker to previous object edge

6 Playback marker to next object edge

7 Playback marker to previous marker

8 Playback marker to next marker

9 Show all

10 Show all vertical

11 Zoom to range

12 Section as zoom step

13 Zoom out of waveform

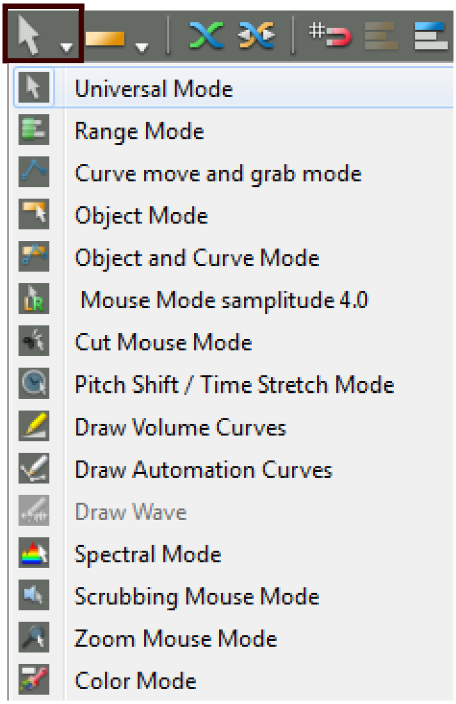
14 Zoom into waveform

15 Overview mode

16 Spectral Display

17 Waveform display

Mouse mode



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

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

Universal mode

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

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

Clicking makes the following mouse functions available to you:

Mouse functions for the upper half of the track:

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

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

Mouse functions for the lower half of the track:

Selecting objects

- Selects individual objects by clicking
- Multiple objects can be selected by holding down the Ctrl key and clicking on them.
- All objects between two selected objects can be selected by holding down the Shift key and clicking on them
- Stretches a rectangle by dragging the mouse pointer to the right. All objects touched by the lasso are marked (see "Lasso function" (view page 104))

Moving objects

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

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

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

Duplicating objects

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

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

Lasso function

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

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

Setting object volume and length with the object handles

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

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

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

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

Editing automation curves in universal mouse mode

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

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

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

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

- If you want to change the vertical positions of the curve points – that is, their values, not their positions – hold the **shift key** while moving them.

- Holding the **Alt key** in addition to this will only move the curve points horizontally. (So you'll be changing their positions and not their values).

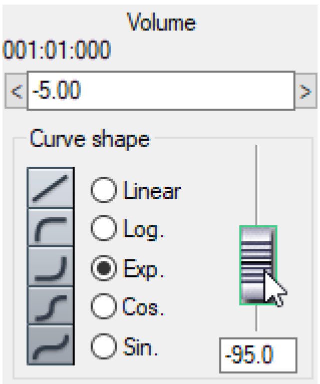
Moving curve segments: If you move the mouse pointer over a curve (the mouse pointer becomes a double arrow) and drag on the curve, both handles to the right and left of the mouse are selected and moved together.

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

Automation points lasso: Clicking and dragging next to the object activates the object lasso (view page 104) for selecting multiple automation points.

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

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



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

Under Waveform you can define how the curve runs from the selected curve point to the next. The possible curve shapes correspond to the curves for objects fade in and out (view page 153). Using the fader you can adjust the precise settings of the curve form of the fade.

Automation curves may also be edited in universal mode. Learn more in the "Automation" chapter.

Range mode (safe mode)

Right mouse button: Displays context-sensitive popup menu.

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

Create ranges

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

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

Setting playback markers

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

Zoom

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

Temporary switch to other mouse modes

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

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

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

Curve mode

This mode is strictly for editing automation curves (view page 428). It can be opened temporarily from "Universal/Range" mode by pressing and holding down the hyphen ("-") key.

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

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

Left mouse button:

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

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

Clicking on a point: Selects a point.

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

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

Dragging a selected point: Moves the point.

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

Right mouse button: Context menu

Object Mode

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

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

Left mouse button:

Click: Selects objects.

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

Click + Ctrl: Selects multiple objects.

Dragging objects: Moves objects in the set grid steps.

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

Drag + Ctrl: Duplicates one or several objects.

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

Double-click on objects: The object editor opens.

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

Object handles:

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

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

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

> Detailed information about how to assign keyboard shortcuts for temporary functions is provided in the menu reference, help file, or in the "manual.pdf" documentation under "Menu File -> Program Preferences -> Edit Keyboard Shortcuts and Menu (view page 626)".

Object/Curve mode

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

Left mouse button:

Click: Selects objects.

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

Click + Ctrl: Selects multiple objects.

Dragging objects: Moves objects in the set grid steps.

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

Drag + Ctrl: Duplicates one or several objects.

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

Double-click on objects: The object editor opens.

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

Object handles:

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

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

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

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

Clicking on a point: Selects a point.

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

Dragging a point: Moves the selected point.

Dragging a point + Ctrl: Moves several selected points.

Right mouse button: Context menu

Left/Right Mode

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

Left mouse button:

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

Right mouse button:

Click: Selects objects.

Click+Shift: Selects multiple objects.

Dragging objects: Moves objects in the set grid steps.

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

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

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

Clicking on an automation point: An automation point is selected

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

Dragging selected automation points: Moves selected automation points

Cut Mode

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

Under "System Options > Keyboard, Menu & Mouse > Special Keys > Temporary key for cut mode (view page 626)", you can specify a keyboard shortcut to keep "Object Cut" mode active provided the shortcut key is held down.

Right mouse button: Context menu

Pitch shift/Timestretch mode

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

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

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

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

The time stretch mouse mode simplifies working with tempo markers in the timeline. Create BPM markers (tempo changes) by directly clicking the desired playback position and holding down the Shift key. You can adapt the tempo "in one go" by moving the mouse vertically.

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

Right mouse button: Context menu

Volume Draw Mode

This mode allows volume automation curves (view page 428) to be drawn with the left mouse button.

Automation draw mode

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

Detailed information about automation is available in the chapter Automation (view page 428).

Left mouse button: Freehand draw function for volume curves.

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

Right mouse button: Context menu

Wave draw mode

If you are in audio editing mode (view page 66), you can edit the waveform of a file using the pencil tool (view page 145). The waveform display shows a suitable zoom level and the mouse pointer becomes a pencil. Changing the waveform in the wave window is useful if you want to remove very short impulse disturbances manually.

Spectral Mode

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

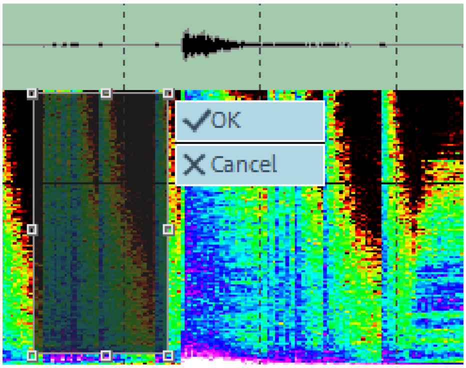
To work with "Spectral Mode" effectively activate the Spectral View in View Options.

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

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

A continuous sound is displayed by a pattern consisting of horizontal lines, which correspond to the sound components or overtones of the sound. A distortion with an impulse quality is seen as a vertical line.

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



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

Left mouse button: Marks the selection box

Right mouse button: Context menu

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

Scrubbing Mode

This mouse mode enables you to monitor and control the playback speed (view page 143). In this case, the project can be played back in a forward or reverse scrubbing direction.

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

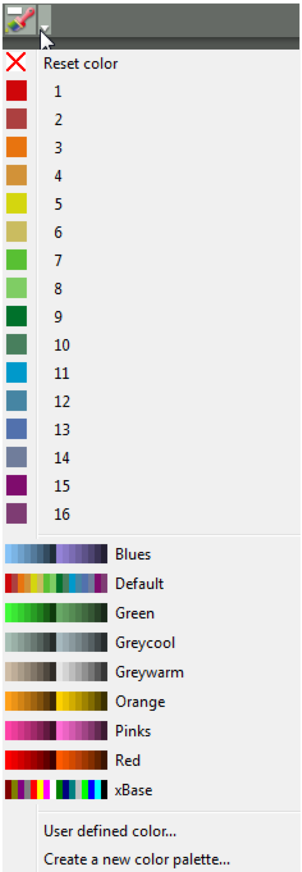
Right mouse button: Context menu

Zoom Mode

Use the right mouse button to zoom out of the project and the left button to zoom into it (view page 121).

Color mode

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



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

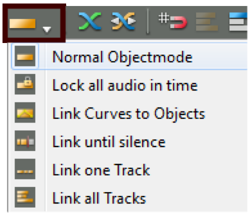
color to be assigned to a track. If multiple tracks are selected (view page 139) beforehand, the color selection will apply to all of these.

Left mouse button: Changes the background color of the object

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

Object modes

The object mode determines the behavior of objects while moving.



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

Normal object mode

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

Lock all objects

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

Note: You can lock individual objects against moving by clicking the lock symbol on the object. It is still possible to change the start and end time then. In the System Options under Program > Object Lock Definitions (view page 625), you can specify against which changes exactly locked objects should be locked.

Link curves to objects

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

Link until silence

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

Link one track to the right

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

Link all tracks to the right

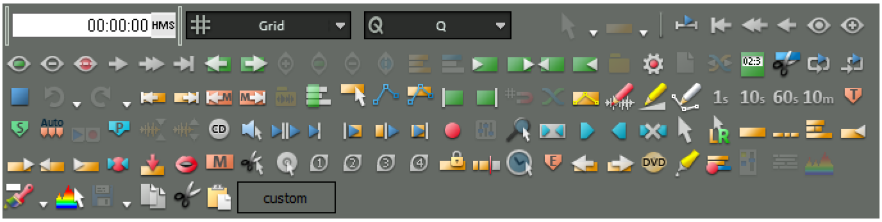
In this mode, all objects in the project that are to the right of the clicked object are moved together with it.

More information about using the object modes is available in the chapter "Working with object modes (view page 158)".

Customize Toolbar

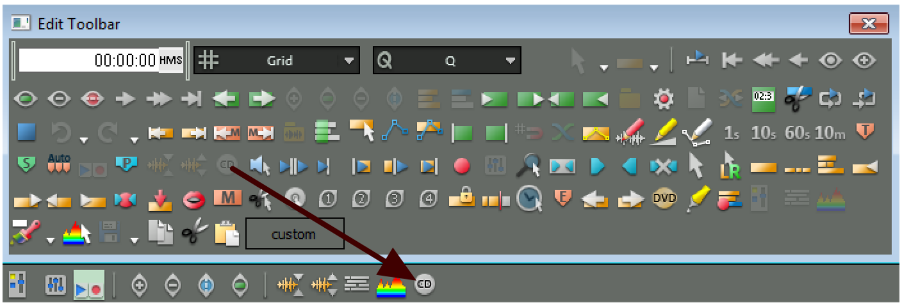
You can customize the toolbars to fit with your requirements by adding icons, changing their positions or removing the ones you don't need. To do this, right-click to open the context menu and select the option "Edit Toolbar".

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



The icons that area already on the toolbar and grayed out in the display. The others can be added by dragging & dropping them onto one of the toolbars.

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



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

Working in the Project Window

Here you will learn how you can work efficiently with Samplitude.

Working in Virtual Projects

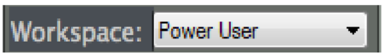
- The keyboard shortcut "A" produces a **range that covers everything**.
- To set a **range between two markers**, double click somewhere between the markers in the marker list.
- The keyboard shortcut "Shift + mouse movement" may be used to **move the range**.
- Drag the range border to **increase the range size to the left or to the right**, for example to adjust a loop.
- Use the "Tab" key to switch **between two display modes** in virtual projects, and edit the modes via "Shift + Tab".
- The function "Objects -> Set objects" enables **objects to be locked against unintentional moving**.
- By pressing the right mouse button in various areas of the VIP, e.g. above an object, above the "Record" and "Play" buttons, or above the scroll bars, a **context menu** with useful functions appears.
- Press the "Delete" key to **delete selected markers and objects**. If you drag out a range over objects, press the delete key to **delete the contents of the range**.
- Press "T" to split an **object at the position of the playback marker**. If a range over an object is selected, cuts will be made at the edges of the range. If "Auto crossfade" mode is active, a crossfade will be added to each cut procedure which may be optimized later with the crossfade editor.
- "Shift + mouse-click" on the 4 zoom buttons S1 to S4 at the bottom of the position list may be used to **save each separate zoom setting**.
- Double clicking an object in the VIP opens the **object editor**.
- You can **select** multiple **objects** one after the other by clicking the lower half of the objects while holding down the "Shift" key.
- When holding down "Shift", **all objects located between the first and last activated object** will be **selected**.
- There are a number of functions for **editing objects** available via "Object -> Edit object/crossfade", which can be executed quickly by clicking on the corresponding keyboard shortcut.
- The **level and panorama curves** can be **linked to the objects below** by pressing the "Connect curves and objects" button in the mouse mode bar. This way, objects and the curves that belong to them can be moved together.
- **Numerical values** in dialog fields like the display of the playback position in the transport window can be changed by **clicking the corresponding field and holding down the left mouse button while moving up or down**. This turns the mouse pointer into a double arrow.

Workspaces

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

For this you can hide certain menu items (menu "File" > "Program Preferences" > "Edit keyboard shortcuts and menu... (view page 626)") > "Hide Menu Item") and customize toolbars (right mouse click on the toolbar) so that they only contain the icons and menu items that are required for a specific task such as mastering, editing or recording. The arrangement, visibility and docking status (see below) of all windows in the Workspace are also saved.

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



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

Docking

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

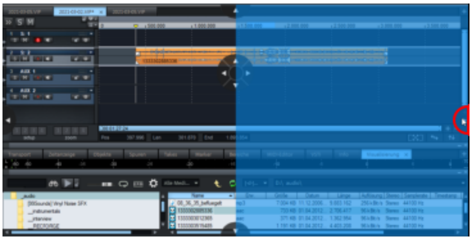
Each window (projects, manager, visualization, object editor, MIDI editor...) can exist on the surface as undocked ("floating") or docked. Undocked windows can be placed on top of other windows, can be freely resized and have a title bar. Docked windows share a screen area with other windows.

Docking windows

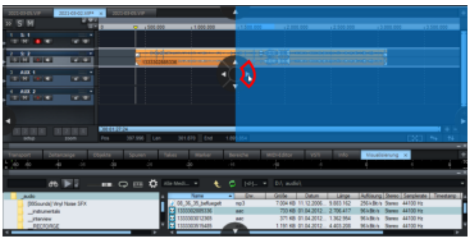
You can dock undocked windows by touching them in the title bar and moving them to the designated areas on or in other windows. While moving, arrow icons will appear on the surface of the corresponding target window to determine the docking position.

The docking areas have the following meaning:

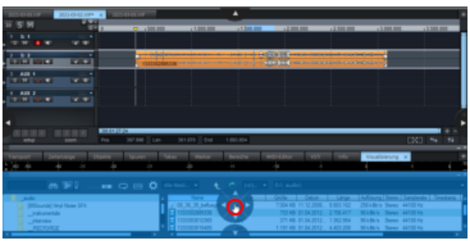
Dragging on one of the outer arrows will dock the window outside of the existing windows, thus creating a whole new docking area.



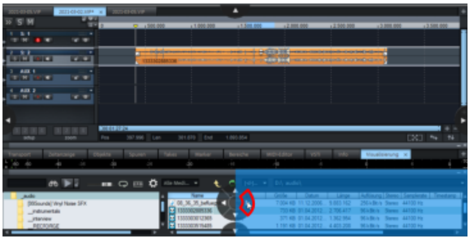
Dragging on one of the inner arrows will dock the window in an existing docking area, the two windows will then share the screen space previously occupied by the window alone.



When dragging on the circle in the middle, the windows are superimposed. They now occupy the screen area together and are grouped in a **Docker**.

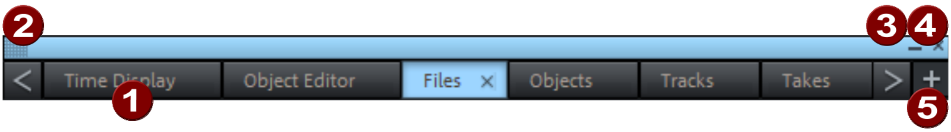


You can also place and arrange new windows inside a docker itself. To do this, simply move the respective window using the mouse onto one of the arrow symbols in the docker.



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

3. 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.

4. Close the Docker with the close button. The Manager/Docker can be shown and hidden in the **View menu**.

5. 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 have already been integrated appear with a (+) symbol. Entries for windows that can only exist once in the program are grayed out in the menu in this case.

> If a Docker was formed by combining project windows, the + button opens the **New Virtual Project** dialog.

Two Dockers are permanently available in Samplitude:

- All managers and many other dialog windows are opened by default in **Manager/Docker**
- All project windows are combined as new tabs in a Docker.

Use the shortcut keys Ctrl + B to switch to the Manager/Docker, Ctrl + P to switch to the Project Docker. You can then use Ctrl+Tab to switch between tabs within a Docker at the touch of a button. Use the Ctrl + F4 shortcut to remove the currently selected tab without opening it as a separate window.

Zooming

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

Samplitude features the following zoom functions:

Zoom with the Position Bar

Click the magnifying glass to open the zoom feature. The glowing red buttons zoom vertically along the timeline while the blue magnifying glasses zoom vertically. In addition, there are **4** freely definable magnifying glasses; pressing a number and holding down "**Shift**" while clicking enables the zoom stage to be set individually.

Use the wave symbols on the right to set the zoom stage for display of the waveform in the active section.

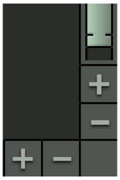
Zooming with the keyboard

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

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

Zoom using buttons

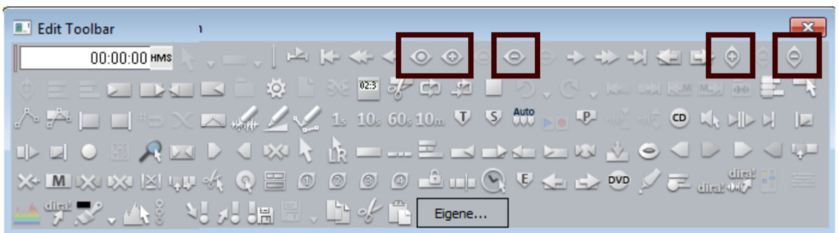
In the lower right-hand corner of the VIP window, the "**+**" and "**-**" buttons may be used for more precise adjustment of the horizontal and vertical zoom levels.



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

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

You can add additional zoom buttons to the context menu via the "Edit Toolbar" dialog. (view page 115)



Zooming with the mouse wheel

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

Shortcut keys for zooming with the mouse wheel

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

Zooming with the Mouse - Vertical Zoom

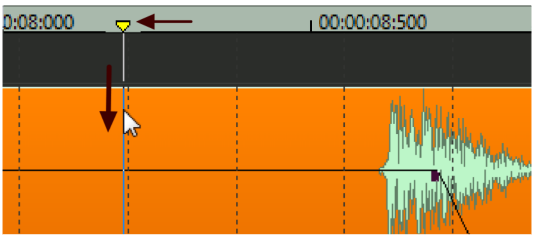
An elegant method of zooming is to left-click on the time line, hold down the button, and drag the mouse vertically.

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

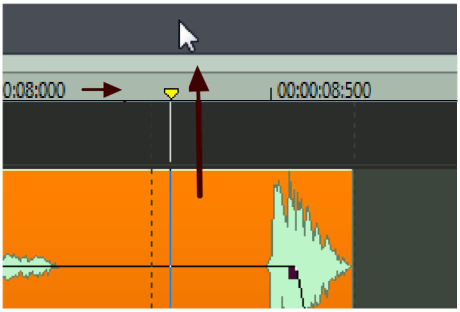
This allows you to position the playback marker precisely "in one go".

Step 1: Click on the the marker bar to roughly position the playback marker.

Step 2: Next, move the mouse downwards while holding down the mouse button to zoom into the project.



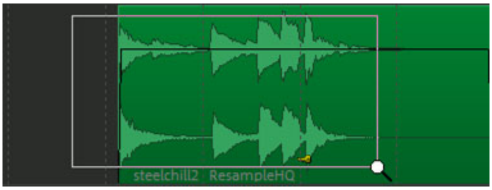
Step 3: Correct the mouse position by moving it horizontally and then zoom out of the project by moving the mouse upwards.



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

Zoom Mode

You can switch to Zoom Mode (view page 112) by clicking the magnifying glass icon (Zoom Mouse Mode). The mouse pointer will turn into a magnifying glass.

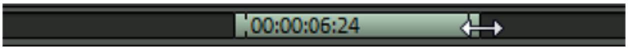


Use the right mouse button to zoom out of the project, and click with the left button to zoom into it. In this mode you can also draw a rectangle around the area you want to zoom in on.

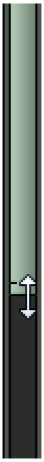
The Zoom mouse mode can also be temporarily activated by holding the Z key.

Zooming with the Scrollbars

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

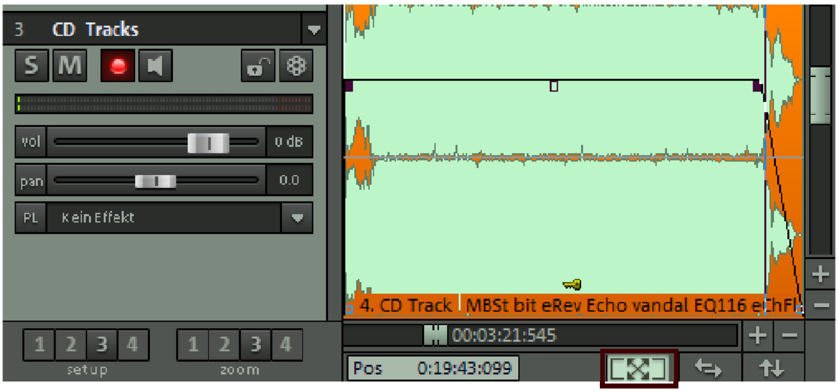


Similarly, you can use the vertical scroll bars to zoom into the track display.



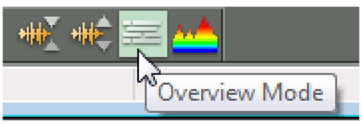
Zoom with the button

With the „Zoom to selected object“ button the displayed project section jumps to the selected object. Clicking the button again resets the display to the previous view.

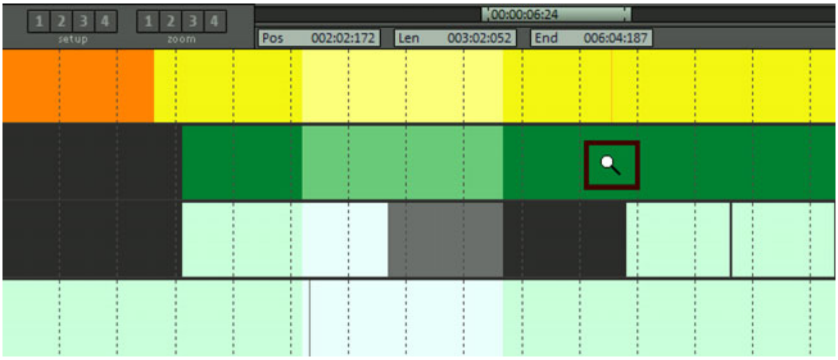


Zooming with the Overview Mode

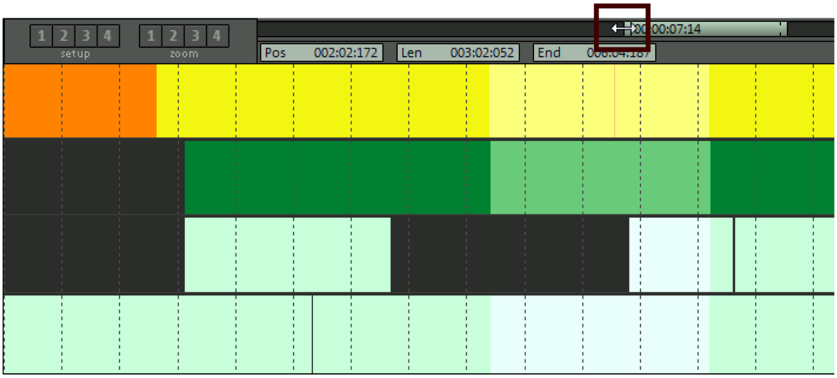
Overview Mode offers you a view of all the objects in your project in a clear display underneath the arranger track. You can activate overview mode with this button



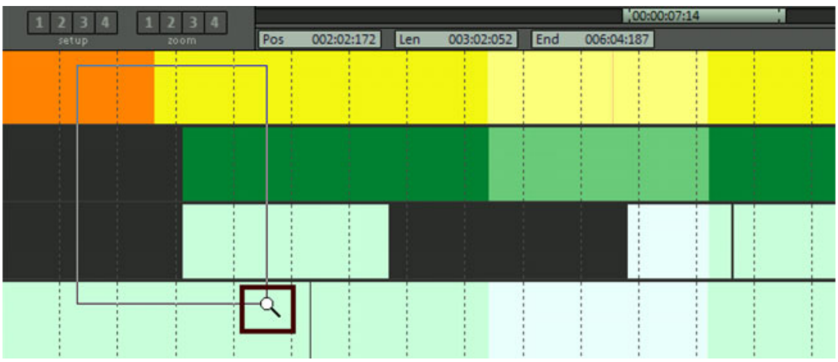
or with the corresponding command in the "View" menu. The mouse pointer turns into a magnifying glass in this screen area. The section chosen in overview mode determines the display size and the position of the section shown in the arranger. You can adjust the section in the arranger by clicking on the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



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

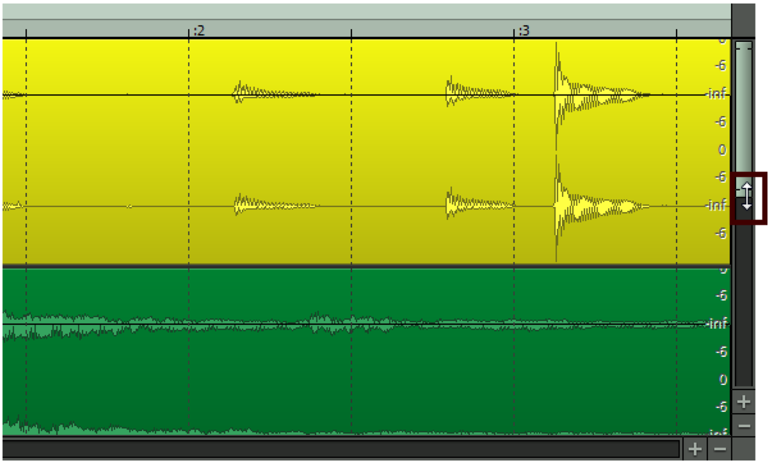


To select the desired section, drag out a lasso around the overview using the mouse.

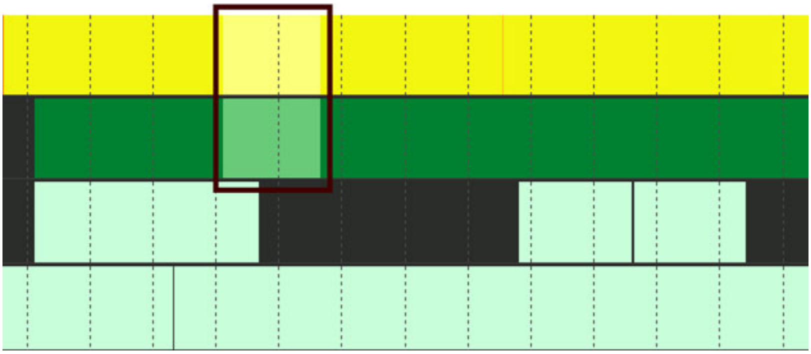


If another position in the overview is selected or a new lasso is drawn by left-clicking, the horizontal (i. e. temporal) position in your arrangement will change.

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



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



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

Detailed information about creating keyboard shortcuts can be found in the menu reference under „File“ > „Program Preferences“ > „Edit Keyboard Shortcuts and Menu“ (view page 626).

Scrolling

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

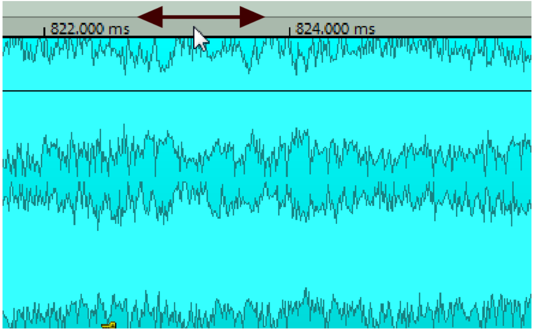
The visible section follows the playback marker during playback in what's known as Autoscroll Mode. This means that the section "jumps" to the next time section shortly before the playback marker moves off the right side of the screen (Page Mode). In "Soft Autoscroll" mode, the section moves constantly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

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

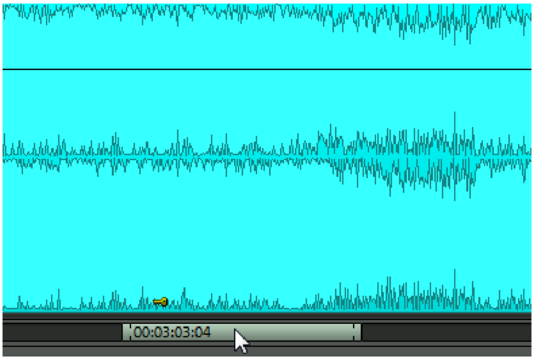
The Home and End keys move the playback marker and the visible section of the project either to the very beginning or the very end.

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

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



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



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

Tip: Other scroll buttons can be added (view page 115) in the "Edit Toolbar" dialog.

With keyboard: Under "View" > "Horizontal/Vertical" (view page 1036) you'll find a range of scrolling commands that can be carried out using your keyboard. Set up keyboard shortcuts for menu scrolling commands to be able to navigate (view page 538) through the arrangement quickly using keyboard commands.

Working with Ranges

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

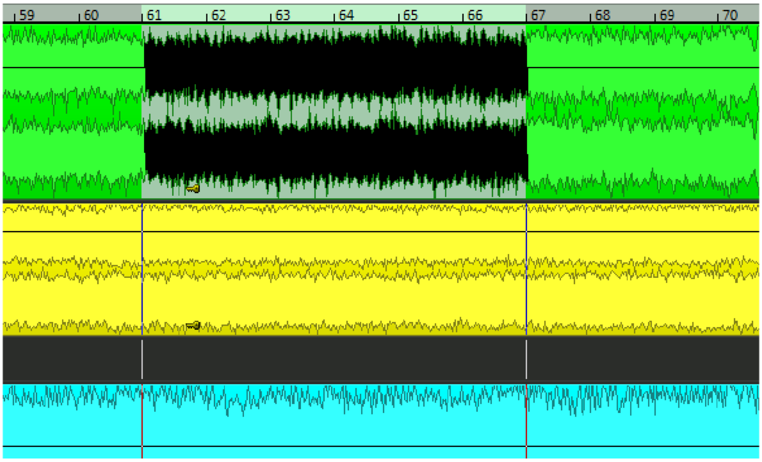
Selecting a Range

Step 1: To select a range, set the mouse mode (view page 102) to "Range Mode" or move the mouse pointer to the top half of an object and press the left mouse button.

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

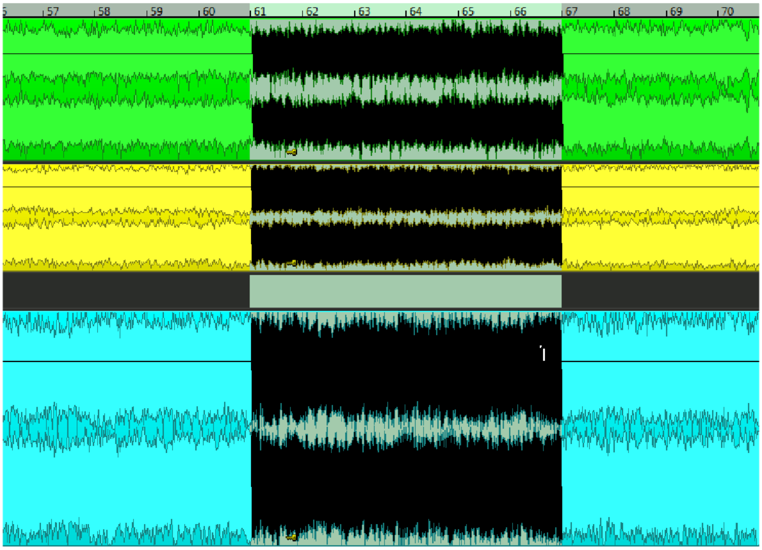
If you have activated the grid function, the range will automatically be adjusted to the set grid value.

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

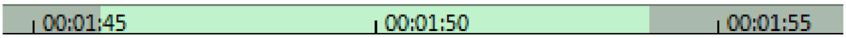


The playback marker always automatically remains at the beginning of the range, even if you can't currently see it.

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



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



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

It is important to note the following when doing this:

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

Leaving a Range

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

Reactivate Range

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

Change Range Borders

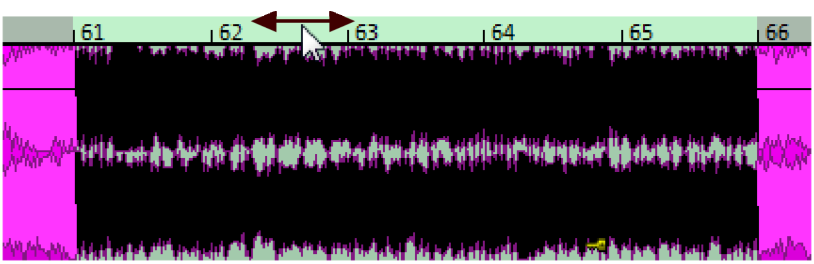
You can change range borders by placing the mouse pointer on the edge of a range in the grid bar until it turns into a double arrow. Now you can reset the range borders by dragging in the horizontal direction.

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

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

Horizontal Movement of a Range

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



Saving and opening ranges/Special range commands

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

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

> **Note**: Alt + F4 is excluded since it is a Windows shortcut for closing windows. Similarly "Alt + F9" should not be used either, as it is used in Sequoia as a source destination edit command. However, you can redefine this keyboard shortcut any time you want via "File" > "Program Preferences" > "Edit Keyboard Shortcuts and Menu" (view page 626).

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

All saved ranges are listed clearly in the Range manager (view page 188) and can be managed there.

Working with Ranges: Examples

Moving several neighboring objects together

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

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

Removing a section of a song from the project

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

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

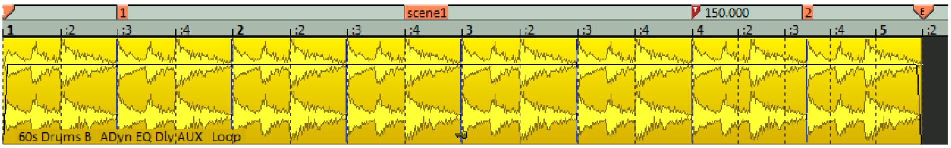
Step 2: Now select "Edit" > "Delete" > "Ripple delete" to remove the verse.

> Additional special functions for defining, changing and using ranges can be found in "Edit" > "Range".
>
> All saved ranges are displayed In the Range Manager (view page 188) where they can be numerically edited and recalled. The Range Manager can be found in the docker or opened through "Edit" > "Range" > "Range Manager...".

Under "Edit" > "More" you will find an additional dialog window called the Range Editor (view page 668) which can be used for editing the range start, range length and range end.

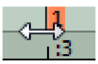
Working with Markers

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



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

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



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

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

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

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

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

The Marker manager (view page 184) can also be accessed through "View > Manager > Marker manager". This is where you manage and edit the markers you set.

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

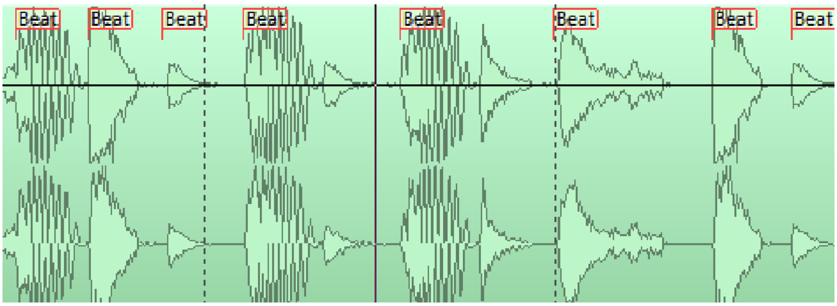
Use the function keys F2 and F3 or the shortcuts Alt + W/Alt + Q to quickly jump back and forth between the marker positions or select "Play/Rec > Move playback marker > Marker left/right" from the menu.

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

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

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

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



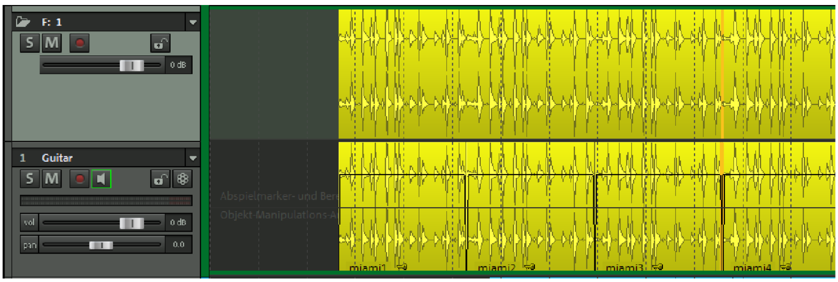
Note: All time information for the audio marker/beat markers relates to time positions in the audio material and not to positions in the virtual project.

- **The triangular markers for CD burn functions**: CD track markers are displayed in red, CD subindex markers are green, and CD pause markers are blue.

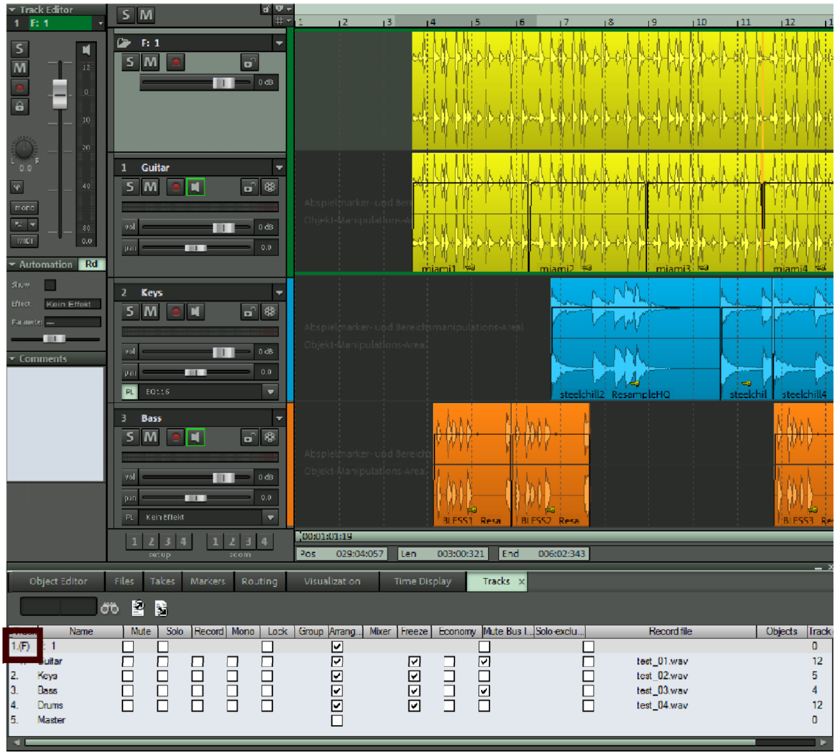
- **Tempo markers** (view page 396) signify a tempo change at a specific position in the project. Double-clicking a tempo marker opens the "Tempo and Time Signature" dialog.
- **Bar markers** (view page 398) change the time signature starting at the marker position, e. g. 4/4 beat to 3/4.
- **Grid position markers** (view page 398) assign a specific musical position to a specific time position. This way, the beat grid and MIDI events can be easily synchronized with existing audio material.
- **Lyrics markers:** Lyrics markers enable you to enter song text, comments or production 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.
  - In the score editor (view page 369), the lyrics markers can be displayed as song lyrics on the notepad.
  - Optionally, lyrics markers can also be displayed on the marker bar (in the Program settings > VIP Display (view page 631)
  - The time display (view page 1009) can be configured to display the name of the current and following lyrics markers in a field.

Folder tracks

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

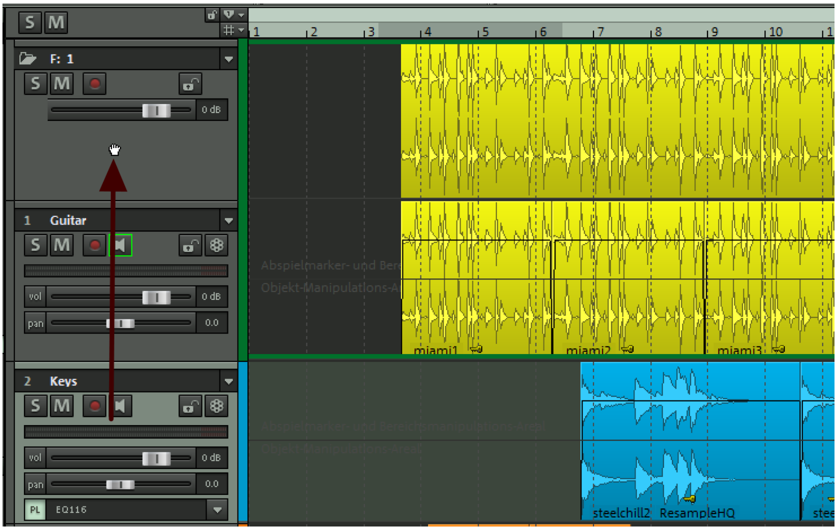


The tracks in the folder track are displayed within a frame and the folder track number is appended with an "F" in the track manager.



If you want to delete a folder track, a dialog will ask to confirm if you really want to delete all of the tracks contained in the folder as well.

You can drag & drop tracks into the folder track by clicking on the track name or an open area in the track header. The mouse pointer turns into a hand. Now you can drag & drop the track into the track header of the folder track.



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

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

- Mute, Lock, and Solo

All tracks in the folder track are displayed when the folder track is open. Each track is given a border with the track color of the folder track. When minimized, the tracks found in the folder will be hidden in the arranger. They will still be visible in the mixer. Tracks of a folder track are marked with the border color of the folder track in the mixer. Normally all tracks can be seen in the folder track along with their objects in a minimized state.

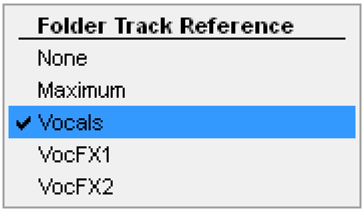
Range selection in the folder track is carried out across all tracks. You can use this function for range-based editing across all tracks in the folder track.

Reference Track in the Folder Track:

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

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

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



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

Select multiple tracks – group track controls

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

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

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

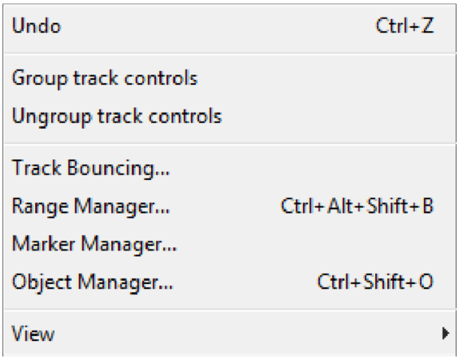
On the other hand, specific settings such as integrated plug-ins or the comments field remain as individual settings for the respective track and are not transferred to the group.

This group functionality also applies to the mixer. The multitrack selection function works additionally to previously added control groups (view page 219) of the mixer if the same control elements are used.

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

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

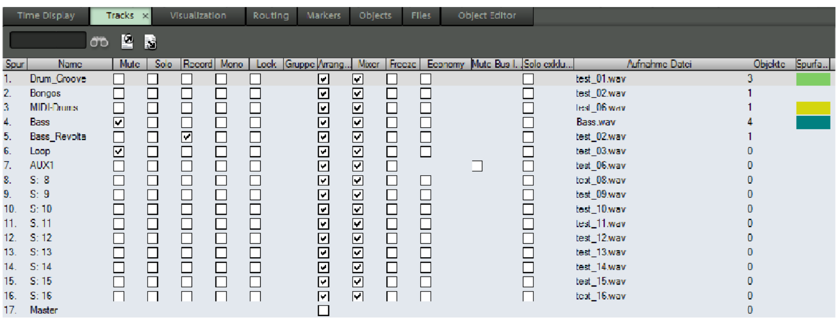
The context menu featuring commands to group or ungroup track controls can also be accessed by right-clicking in the track header area below the last track in the arranger.



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

Hide tracks

If the project window contains tracks that are not being used, you can hide these tracks and mixer channels to create a less cluttered interface. To do this, open the Track Manager (view page 182)(menu "View" > "Manager" > "Track Manager") and deactivate the corresponding checkbox in the columns "Arrangement" and "Mixer" to hide the tracks.



With the key combination Ctrl + left click or Shift + left click you can also select several tracks and hide them together.

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 and mixer). Select one or more tracks and then select one of the options:

- **Select all track routing destinations:** 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.

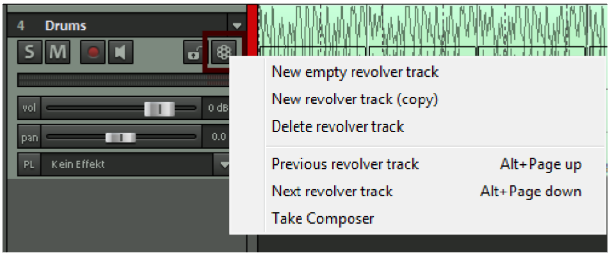
You can execute these commands before the following ones, e.g. to hide tracks together with their buses or to display only the tracks that are routed to a certain bus.

- **Show only selected tracks:** All unselected tracks are hidden.
- **Hide selected tracks**
- **Unhide all tracks**
- **Synchronize Mixer and Arranger:** The visibility of the tracks in the Mixer is adjusted to that in the Arranger.

Revolver tracks

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

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



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

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

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

To edit or view existing revolver tracks in the overview, open the Take Composer (view page 192) for a particular track. This editor enables revolver tracks and object takes to be edited, and the created revolver tracks are displayed under the individual takes.

Advanced Ruler / Time Display

Use the grid bar's context menu to activate a second grid line in the arranger and set your own measurement unit (Show 2nd Grid Line). Both grid bar positions can be exchanged (Exchange Grids). If "Independent units" is selected in the transport control, (view page 90) the upper grid will not adapt to a chosen BPM grid (view page 401) like the lower grid does. This makes it possible to select an SMPTE format independent of the project frame rate, which provides an overview of two different SMPTE displays.

Scrubbing

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

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

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

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

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

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

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

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

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

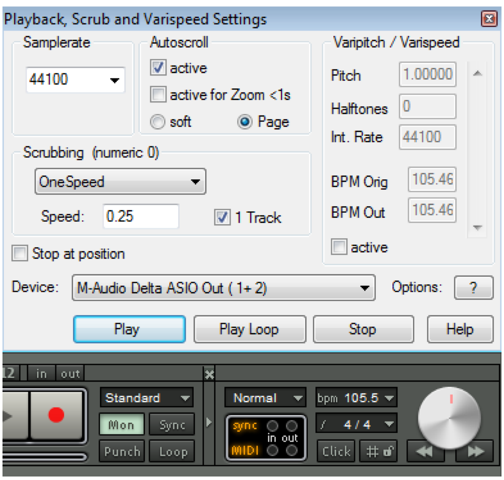
One speed: The preset scrubbing speed is 1.0, i.e. original speed. The "Shift" key halves this value. The "Ctrl" key uses the speed set in the "Scrubbing speed" field.

Scrubbing speed: This specifies the factor of the original speed applies to the scrubbing speed. The range of values stretches from 0.01 to 10.0, i.e. 1/100 of the original speed up to ten times the regular speed.

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

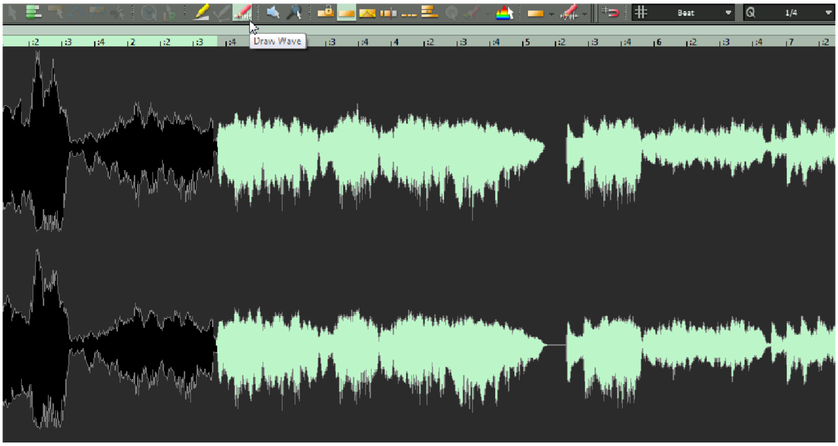
Scrubbing in the transport control

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



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

Drawing Waveforms with the Pencil Tool



In "Wave editing (view page 66)" mode, use the pencil tool to draw or modify the waveform of a file. Waveform display shows the suitable zoom level, and the mouse pointer becomes a pencil. Changing the waveform in the wave window is useful if you want to edit very short impulse disturbances manually.

Object Editor

The foundation for object-oriented working is the object editor which can function as a tool for a single object or as a "realtime channel strip" for multiple objects.

Note: There is an object editor for MIDI objects as well. Detailed information can be found in the chapter "MIDI in Samplitude > MIDI object editor (view page 326)"

The object editor is divided into three sections – "FX", "Fades", and "Time/Pitch". The object editor can always be left open to edit objects - the view automatically adjusts to the selected object.

To open the object editor, select an object and press "Ctrl + O" or double-click on the object. The object editor opens in the Docking bay (view page 90).

Object Editor: Basic Functions

The selection options for the three different dialog windows of the object editor are displayed on the left.

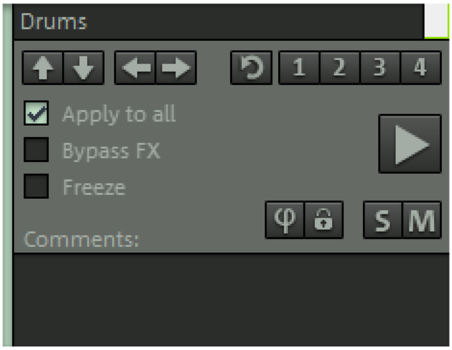
FX: Here you can set the gain, AUX, plug-ins, EQ, panorama and automation.

Fades: Here you can set the fade-in, fade-out, wave position and object position.

Time/Pitch: Here you can set time stretch / pitch shift algorithms and loops.

Note: By right-clicking in an open area you can display and select various graphic interfaces which are also known as "skins". The skins that have the "Editor Max" in the name display all of the parameters in a single overview.

The right section is the same in all three dialog windows:



The object name is displayed in the upper text field and can also be edited. In the square field beside this you can select the color for the object waveform.

Note: Please make sure that you have selected the option "Default color settings" in the "Waveform Colors" section of the display options (keyboard shortcut: Shift + Tab).

Using the arrow keys you can "jump" up or down to the object of a neighboring track. With the horizontal arrow keys you can jump to the previous or next object on a track. If multiple objects are selected, these buttons will not be active. Clicking the Undo button (spinning arrow) returns the selected object to its original settings.

1-4: You can save the various object editor settings in the 4 snapshots. This can be done by pressing "Shift" and left-clicking on the corresponding button. If you haven't yet set a snapshot for the respective button, a simple left-click is enough.

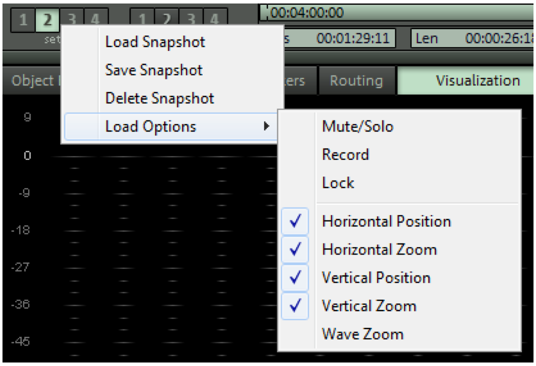
Clicking on the respective number keys will switch between snapshots. This means that you can quickly compare several object editor settings.

The active snapshot is marked by a bullet point.

To apply object settings from one object to another, first save the object settings from the original object as a snapshot, then switch to the next object and open the snapshot for the currently selected object by simply clicking on it.

Right-clicking on a snapshot number opens a context menu where snapshots can be saved, loaded and deleted.

In the context menu you can also save the object effect settings as a file or load them from a different file. When loading a file you can also specify exactly which settings from the snapshot you want to apply to the activated object. Here you have a choice between volume, gain, panorama, EQ, AUX and plug-in settings.



To all: If the "To all" option is active all selected objects are updated when the object editor is opened. No matter in which of the three object editor windows, all recently applied settings that have been made in a selected object, will be applied to all other selected objects.

Note:

Volume changes are transferred relatively in an object, i. e. they are increased or decreased relative to the volume of other selected objects.

The EQ settings are always completely transferred, in contrast to previous versions where only the edited bands were transferred.

Bypass FX: This deactivates all object effects, EQ/panorama settings, and AUX sends.

Freeze: If you click on the Freeze function for an object, this will be calculated as a new audio file with all the associated effects. The new freeze object takes the place of the original object. This way you can bypass the realtime processing of effects and reduce the load on the CPU.

Play/Stop: This button corresponds to the normal playback function (spacebar).

Phi: The Phi button causes a 180 degree phase rotation. By right-clicking on the Phi button you can invert the left and right channels individually.

Lock: The "Lock" button has the same function as the key button in an object. The object is then protected against any inadvertent horizontal movement. Additional locks against vertical moving, volume adjustment, fades, length adjustments, ripples or deletion can be set in the system options (keyboard shortcut: "Y") under "Program"

> "Object Lock Definitions (view page 625)". Locking can be temporarily cancelled by pressing the "Alt" key.

Solo: During playback this button moves the playback marker to the start of the selected object and plays that only. Track-dependent busses and AUX paths are also included in the playback.

Mute: This button mutes the selected object. By right-clicking you can also mute the left and right channels individually.

Comments: You can type comments relevant to the selected object in this field.

Object effects

In addition to the numerous effects available for individual tracks and the master area of the mixer, audio objects can also feature effects. All settings are retained when moving or copying objects.

By crossfading between objects of different settings, you can also fade between the different effect settings. CPU-intensive effects can therefore be used more efficiently, since they are only processed as required (unlike a track effect set up as an AUX send effect).

"Effects" or the context menu also makes it possible to access the object effects.

Manipulation using the level and pan curves or the volume controllers is only calculated after object level effects.

Gain and AUX sends

Gain: Set the gain for the object here.

AUX Sends: Any object can send to all available AUX busses. You will see, however, initially only to AUX1 AUX 4. You can adjust the corresponding slider by dragging in the respective AUX Send field.

By right click on the blue button in the header the advanced AUX send dialog will open, which provides full access to all AUX send paths.

By left clicking on the blue button in the track header you can bypass all AUX sends (Bypass function).

Plug-ins and EQ

The plug-in section provides you with quick access to installed effect plug-ins on the object level.

Clicking on the empty insert slot opens the plug-in browser (view page 240), where you can load a plug-in to the slot. Click on an occupied slot to activate/deactivate the plug-in. Right-clicking on the slot opens the interface of the plug-in. The arrow next to the respective insert slot opens a menu with various functions. You can ope the plug-in browser again to switch out the plug-in or remove it ("No effect"). The "Plug-ins" button activates or deactivates all effects in the channel. A visual indicator (*) is displayed for plug-ins that were previously active and are active the next time "Plug-ins" button is pressed.

The "FX" button opens the effect routing selection dialog (view page 239) for modifications to the plug-in series. To load additional internal effects, you can use the FX/Routing dialog. If you click on the arrow button next to the "FX" button, you can save, load or reset the object effect settings.

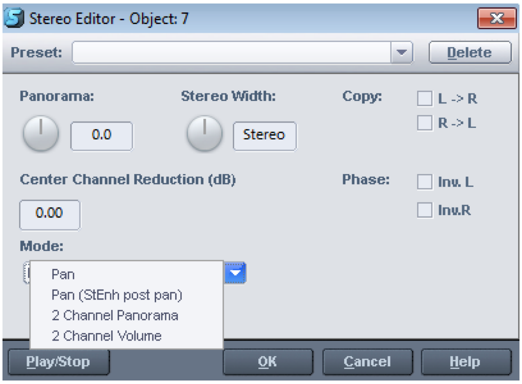
The "Offline-FX" button opens a selection dialog for using offline effects (view page 236). Select the effect in the corresponding effect dialog and confirm using the "OK" button. The effect will now be included in the object offline. It is also possible to edit either the left or right channels individually.

To remove a plug-in or an effect, select "No effect" from the corresponding plug-in slot in the menu.

Under "EQ" you can set four frequency bands of the parametric EQ directly via the knobs. The small square button in the header switches the equalizer on and off. By right-clicking on the knob/button you can access the dialog window for the fully parametric 6-band equalizer. The selected EQ values can be cut, copied, inserted and deleted by right-clicking on the corresponding fields.

Pan

Here you can edit the panorama as well as the stereo width of the active object. Right-clicking on the knob opens the "Pan Settings" dialog.



Use L <-> R to switch between the left and right channels.

Pan: With the knob on the left you can specify the panorama in the stereo image.

Stereo: The knob on the right can be used to set the stereo base width.

Balance + Stereo Enhancer: The pan knob sets the stereo balance. The mid position corresponds to a level of 0dB on both channels. When the panorama is moved to the left, the level of the right channel is reduced and vice versa.

The stereo button sets the base width of the left attack = 0% = mono via mid = 100% = stereo to right attack = 200% = Enhan. (extreme base widening).

-4.5 dB Pan + St. Enh.: The panorama fader controls the panorama position in such a way that when the levels of both channels are set to the middle it reduces by -4.5 dB. In the outer settings one of the channels is faded out, the other is raised from -4.5 dB to 0 dB through the control range. This compensates for any perceived drop in volume in the mid position. Use this mode to place mono objects in the stereo panorama.

2 Channel Panorama: In this mode the two faders turn into two panorama faders for both channels, just as would be expected on a mixer with mono channels.

2 Channel Volume: In this mode you can control the level of the two object stereo tracks separately just as you would on a mixer with mono channels.

Reset: This button resets the panorama mode and the knob.

Note: For a three-times more powerful frequency-selective setting of the base width you can use the multiband stereo enhancer (view page 836)as an effect insert in the plug-in section.

Vol

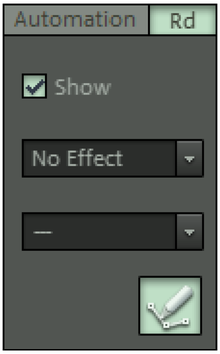
Vol.: The volume can be controlled using this fader.

Double-clicking on the fader sets it to 0 dB.

Norm.: The Norm. button normalizes the maximum level of the object to 0 dB.

Automation

You can change object automation settings here.



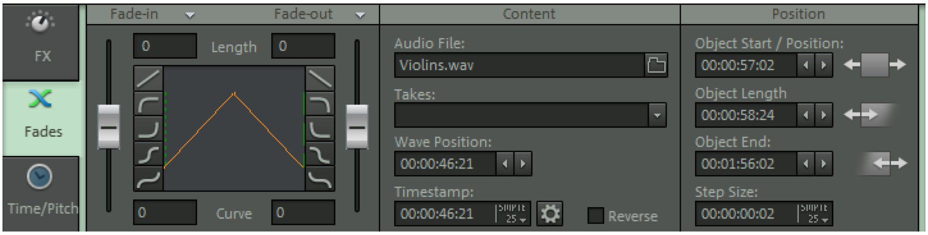
You can make the automation curves of the selected parameter visible in objects with the "View" option.

In both of the lower fields you can select the desired automation parameters.

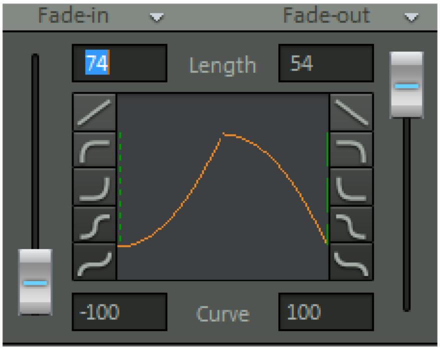
Activate the Automation drawing mode button and draw the object automation curve in the corresponding object.

Position/Fades

Use this dialog to set object-related time positions (start, length, end) and fade parameter for the selected object. In the entry fields "Timestamp" and "Step size" the measurement units can be changed by simply clicking. The following units are available: samples, milliseconds, hours/mins/secs, SMPTE (24, 25, 29.97 non drop, 29,97 drop, 30), SMPTE/milliseconds, CD-MSF, feet and frames 16 mm (40fpf) and feet and frames 35 mm (16fpf), noise reduction (meter).



Fade in/Fade out



Fade in and fade out can be edited in this dialog.

Using the faders you can adjust the precise settings of the curve form of the fade.

If you wish to produce the original fade, click on the arrow button and select the "**Reset**" option.The same context menu also features presets for **"linear"**, **"exp."** (exponential), **"log."** (logarithmic), **"cos."** (cosine) and **"sin."** (Sine) curves. Double-click the fader to switch the fade between "**Linear**" and the last setting.

In the context menu there are two additional options available for fade-ins:

Enable crossfade (left): Use this option to extend the fade-in object to the left over the original object border, depending on the values set in "Fade length". The fade-out of the previous object is shortened accordingly; both fade-in objects will remain linked.

Asymmetrical crossfade (left): This option adjusts the fade-in independent of the fade-out to the left past the object border.

There are also three "Fade offset" option presets which determine which portion of the fade should lie within or outside the borders of the original object.

Fade within: The fade is located completely within the object

Fade symmetric: The fade is symmetric to object borders

Fade outside: The fade is completely outside the object's original borders

"Fade inside" (corresponds to 0% "Fade offset"), "Fade symmetrical" (corresponds to 50% "Fade offset") and "Fade outside" (corresponds to 100% "Fade offset").

The object will be stretched accordingly. The original object edge that now acts as the "axis" of the fade will now be depicted as a dotted line.

If the fade offset is above 0%, make sure that there is enough audio material in the audio file so that the object can always be faded in or out. For example, if the object starts at the exact beginning of the curve display, you will no longer be able to crossfade as soon as the "Fade offset" rises above 0%.

You can also use **Get global crossfade**. Here the crossfade values of the standard settings are acquired for automatic crossfades. Use "**Set global crossfade**" to set the current crossfade values as standard values for automatic crossfades.

The resulting curve form for fade-in and fade-out is shown together with the "Fade offset" in the graphic.

Content

Audio file: Here the referenced audio file can be exchanged, copied and renamed directly in the object editor. Open the file exchange dialog by clicking the folder symbol beside the name of the audio file. If the audio file is used by multiple objects, you can indicate whether it should be exchanged with the currently selected object or with all referenced objects.

Takes: Select the take you want here.

Wave position: The arrow buttons do not affect the position of the object or its length, but instead move the wave display in the object left or right.

Timestamp: This value assigns a unique time to the respective object.

If you click on the "Gear" button the Broadcast Wave Manager will open (view page 602). This feature lets you define and save metadata for the underlying audio files.

The "Reverse" function can be used to play selected objects backwards. This process does not change the audio files.

Note: Please note that there are some limitations when using additional effects on an object that is played in reverse. This applies to audio editing mode or the use of time stretching effects and Elastic Audio.

Position

The starting time of the object on the timeline, object length, and object end can be set here by entering a number or using the arrow buttons.

Using the arrow buttons beside object start / position the object can be moved forward to the start of the project or towards the end of the project.

The arrow buttons beside object length move the start of the object, they work just like the front lower object handle.

The arrow buttons beside object end shorten or lengthen the object, they work just like the back lower handle.

The step size shows the position/length adjustment made upon clicking on the arrow button. The unit of measurement to the right beside the input value can be changed in the context menu (right-mouse click to open).

The lower part of the menu features corresponding presets for each selected measurement, e.g. for the unit of measurement "Bar/beats", the following step sizes are used: 1/64, 1/32, 1/16, 1/8, 1/4, 1/2, 1Beat, 2Beats, 1Bar, 2Bars, 4Bars.

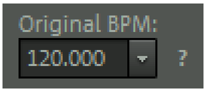
Pitch shifting / Time stretching

Pitchshifting and timestretching can be used in parallel independently of each other, except in the **Resampling** mode.

Pitch shifting: You can shift pitch either in **half tones** (mouse wheel + Ctrl), in cents (mouse wheel + Shift) or enter it as a **pitch factor** of the original value.

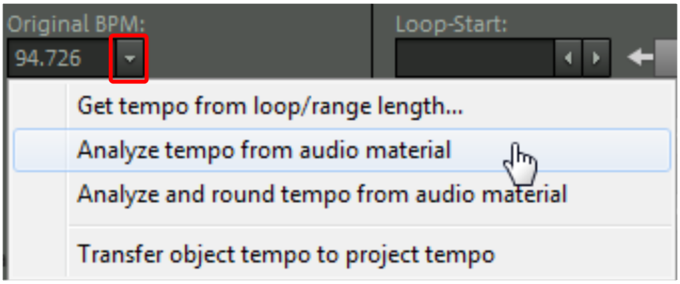
Time stretching: Speed correction can be made either as new object length/**stretch length** or as **stretch factor** of the original value, or you can indicate the desired tempo as BPM (Beats Per Minute).

The original tempo is required to calculate the used stretch factor of the BPM entry.



A question mark next to the entry field indicates that the displayed value is a default value and may not correct.

You can either enter the original tempo manually or let the program calculate it. To do this, click on the arrow next to the value.



Get tempo from loop/range length: A range section above the object is used to calculate tempo. Select the option and enter how many quarter notes should be included in the selection.

Analyze tempo of audio material: Melodyne will calculate tempo. Melodyne must be installed for this. You can learn more in Melodyne integration (view page 292).

Four algorithms may be used for realtime time stretching and pitch shifting: **élastique Pro**, **élastique Efficient**, **Resample** and **Monophonic Voice**.

You can learn more about individual algorithms in the menu reference under "Effects > Time / Pitch > Resampling / Time Stretching / Pitch Shifting (view page 808)".

"Edit" opens a clear display of the already named parameters as well as some others. If during algorithm selection you move through the individual modes with the mouse, a detailed explanation window will appear for each mode.

Elastic Audio provides a high-quality tool for pitch customization. The "Elastic Audio" (view page 810) button opens the activated object in the Elastic Audio editor.

The option **Use Elastic Audio (Pitch Automation)** changes the pitch using Elastic Audio.

If the **Use musical tempo adjustment** option is activated, the object will follow all tempo changes (view page 408).

Loop

In Loop mode (view page 171), Loop Start, Loop Length and Loop End are active whereby the unit of measurement to the right beside the "Step size" input value can be changed with a mouse-click. The loop with the specified step size can be corrected using the arrow buttons:

Loop start: The upper arrow buttons change the loop starting point without changing the loop length, that is, the loop end is moved as well.

Loop length: The middle arrow buttons change the starting point as well as the loop length, the loop end remains unchanged.

Loop end: The lower arrow buttons move the loop end, the loop length changes.

All settings can be seen in the selected object in the form of vertical dashes. Once an object has been selected the whole object turns into a loop. If, however, you selected a range, only this will turn into a loop.

Methods for Working with Objects

Integrating wave files as objects in the VIP

This is how to add a wave file to your virtual project as an object:

- Click somewhere in the upper half of the desired track at the position that should display the beginning of the new object. The playback marker now moves to the position at which the object will be inserted. The track is selected.
- In the "**File**" menu, select "Load/import" and then "**Load audio file**". Browse to the desired wave file in the dialog window that opens. If you now choose to open this, it will be appear as an object in your project window.

If multiple audio files have been selected (by holding down the "Ctrl" key), you can specify under "Options" whether these should be aligned beside one another or on top of one another in multiple tracks and also in which order they should be inserted into the virtual project.

Tip: If you have not opened a virtual project (VIP), the command "Load audio file" will load your wave file in wave editing mode. You can select destructive wave edit mode from the menu, "File" > "Project properties"

Detailed information about destructive editing is available in "Working in the project window -> Samplitude as a wave editor (view page 66)".

Integrating a Section of an Audio File as an Object

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

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

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

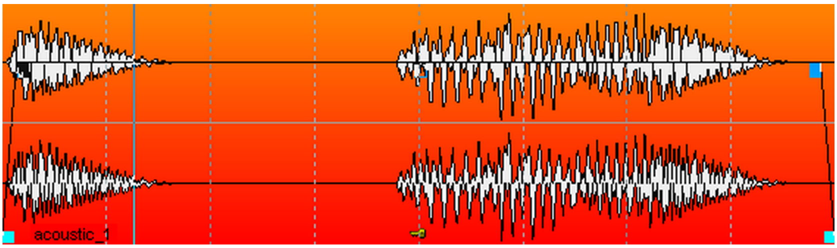
Integrating CD Tracks into a VIP as Objects

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

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

Selecting objects and clearing selection

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



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

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

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

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

- The current object selection may be reversed (invert). To do this, select "Object > Select objects > Switch selection".

Moving Objects

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

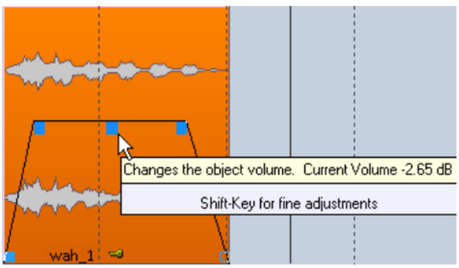
- To move selected objects downwards across the visible section in the arranger, hold the mouse pointer to the lower horizontal scroll bar; after a moment, the arrangement will scroll downwards so that the selected objects can be moved to the tracks below.

Moving Objects in Steps

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

Object volume and fades

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



- The length handles are at the bottom left and right corners. Use these to adjust the length and starting time of an object within the audio file. The maximum length of an object depends on the audio file that the object is accessing.
- The fade handles at the top left and top right fade an object in or out by dragging them horizontally. The curve shape can be adjusted by pulling the handles vertically. You can use the Object Editor to precisely adjust the curve shape.

- The volume handle is at the top center of the object. You can use this to adjust the object volume. The volume level is displayed above the tool tips.
- The object name is visible next to the left length handle on the right hand side and the key symbol to its right can be used for "Locking Objects (view page 164)".

Change waveform view

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

> Detailed information about the display options is available via the menu reference under "VIP display (view page 632)".

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

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

Changing the length and start time of an object

You can move the object borders using the length handles. If the mouse is moved to the length handles at the beginning of an object, the mouse cursor will transform into a double arrow. By clicking and dragging, you can edit the starting time of the object. This also changes the object length.

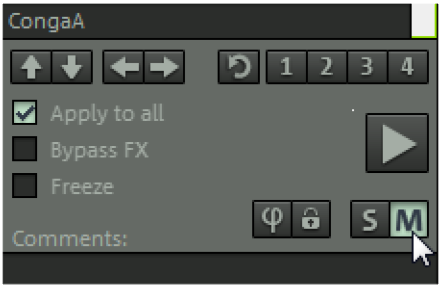
This function is also available in the object editor as well as in the corresponding buttons on the range bar.

Standard operations

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

Mute objects

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



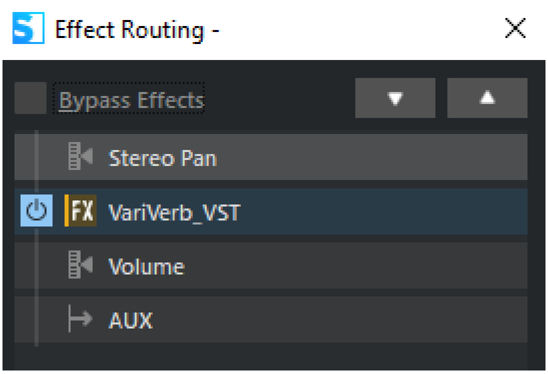
You also have the option to mute either the left or right channel.

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

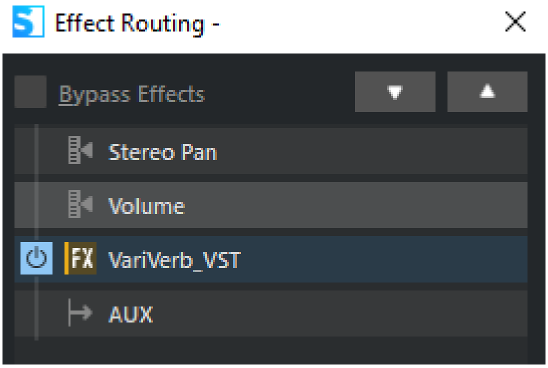
Reverb tails in the Object

If you use echo or reverb effects in the object, reverb tails at the end of an object usually do not sound out. 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.

To change this behavior, open the "Effect Routing" dialog via the "FX" button in the object editor.



For the reverb tail of Room Simulation to decay, change the effect order so that the reverb is placed after the volume.



The reverb time depends on the option "Maximum reverb time for objects without fadeout (sec.)", which is located in the playback options (view page 727).

Locking Objects

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

Clicking on the key symbol again unlocks the object.

Multiple selected objects can be locked with a mouse click.

The menu option "Object > Edit > Lock Objects" applies to single and multiple objects.

If you want to lock all objects on a track, click on the lock symbol in the track overview box or Track Editor.

Locking to prevent changes to fades, length, or deleting can be set in "Object > Edit > Lock Definitions...".

You can temporarily disable the lock function by holding down the "Alt" key while clicking objects.

Offline Editing of Objects

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

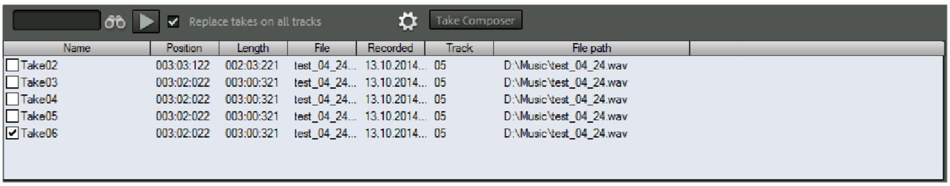
To edit an object offline, right-click the object to access its context menu and select "Wave Editing". This opens the audio file as a "wave project". The wave can now be edited as desired. Samplitude now functions as an audio editor (view page 66).

If an audio file that is used in multiple VIP objects is edited in offline mode using the option "Object > Wave Editing", all of the associated objects in the VIP will access this audio file. For example, if reverb is added to an audio file in offline mode, all objects that reference this audio file will now feature reverb.

However, if you only want to edit specific objects offline, select the option „Object" > „Edit" > „Edit a copy of wave content". Samplitude will create a copy of the audio material and add this to the project folder. The selected objects will now refer to the copy that is created. Offline effects can now be applied to the selected objects without influencing other objects.

The button "Advanced options (view page 757)" in the corresponding effects dialog provides access to options for saving the copy.

The Take Manager (view page 189) provides an overview of the versions of destructively edited files and lets you select them for the respective object.



As a safety measure the option "Create copy" is activated by default for all destructive effects. This creates a temporary copy of the unedited audio file. Activate this option in the offline effects dialog to be able to use the undo function in the VIP later - this allows you to undo offline editing.

Note: Temporary files for the "Undo" function are only created with offline effects calculation if the "Undo" function for wave projects is also activated (keyboard shortcut: "Y" – Program > Undo) and "Create copy" is activated in the corresponding effect dialog. Deactivate the option "Create copy" if you are sure that the original audio file will not be used for other objects. This also applies to objects in other VIPs.

Moving to a defined position

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

1. Right click on the object you want to move. In the context menu that opens, click "Objects/move crossfade/edit -> Move object...". A window opens to enter the

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

Group objects

Select all objects for the group.

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

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

"Object" > "Groups" > "Temporarily exclude object from group" command removes an object from an existing group according to when it was last clicked. By clicking on an object again and repeating this function, the removed object is moved back into the group.

Shortcut: Shift + Ungroup button

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

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

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

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

Linking objects

This is how to link objects in a track:

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

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

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

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

Overlapping objects

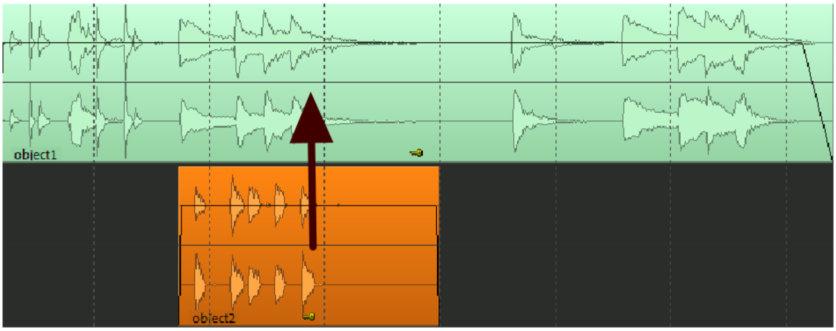
Only one object can be played per track (channel) at any given time.

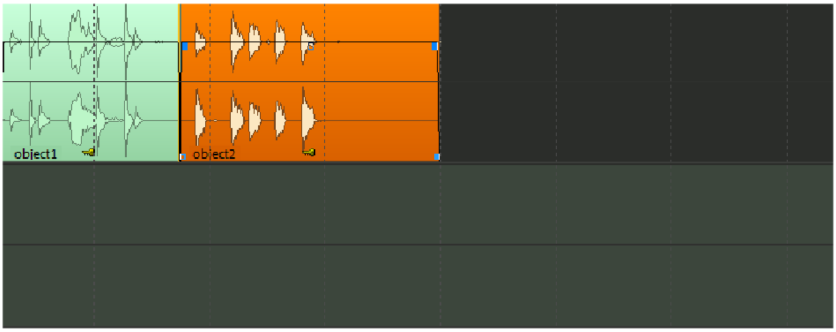
If an object is moved over a bordering object, it will cover the preceding object. The invisible part of the subsequent object can be made visible and audible again by moving the first object. To create a crossfade between two consecutive and partially overlapping objects, apply the "Crossfade editor"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

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

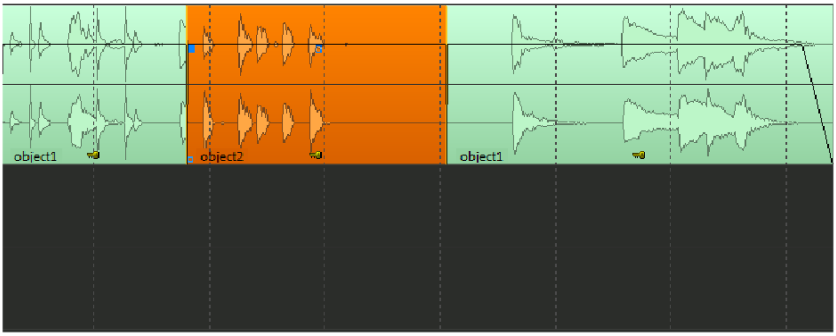
Overlapping objects via "Ctrl + V"

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

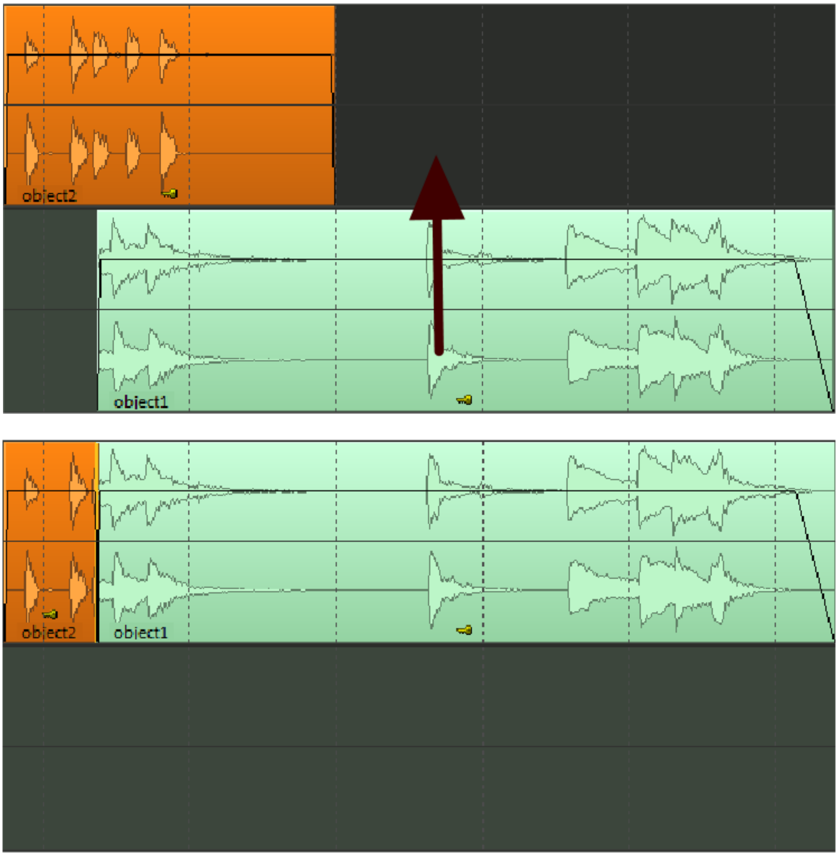




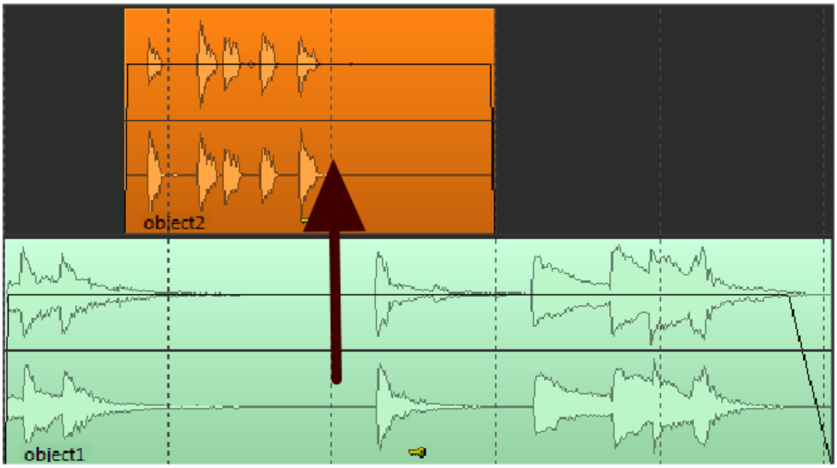
If you hold the keyboard shortcut "Ctrl + V" before releasing, then the shorter object will also replace the longer object across the entire length of the shorter object, but the end of the longer object that extends past will remain.

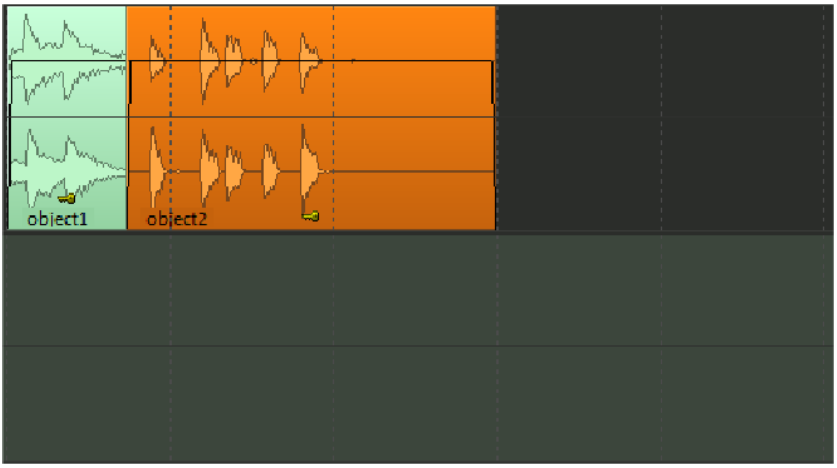


Now try it the other way around and drag a longer object over a shorter object; the shorter object will be replaced from the start position of the longer object if its start position lies behind that of the shorter object.

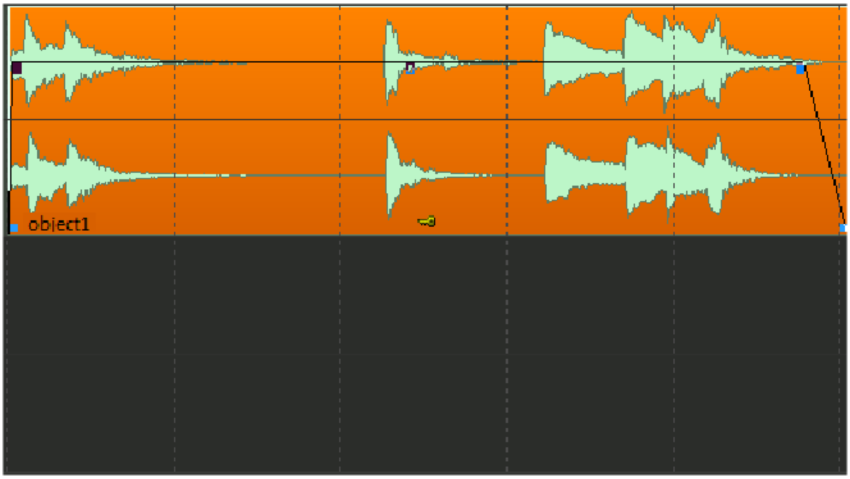


If the start position of the object being moved lies before that of the target object, then the target object will remain complete. Only the part of the object to be moved that is present before the start position of the target object will be added to the target track.





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



Gluing objects

If you want to combine two objects into one, select both objects and right click on one of them. In the context menu that opens, select "Glue objects". This creates a new file that is saved in the "FreezeData" folder in the project folder.

If you uncheck "Freeze" in the object editor of the new object, it will be separated into its old components.

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

Looping objects

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

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

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

An object loop may be edited in far more detail by using the object editor (view page 157). In Loop Mode you can set and customize the loop start, loop length as well as the loop end within the object. Skilled looping omits time-consuming cutting.

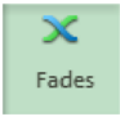
Trimming objects

"Trimming" refers to trimming objects to a certain specific range. Make sure that the selected range does not exceed the object borders.

Once a range is selected, right click the object and select "Trim objects" (shortcut: Ctrl + T). The selected object will now feature the selected length.

You can also trim multiple activated objects in the same way, even if they don't have the same starting and end points.

Replacing an audio file below the object

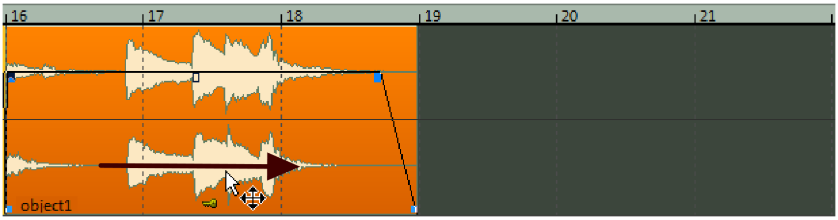


Open the object editor and switch to the "Fades" section or to "Max" view by right-clicking in an open area.

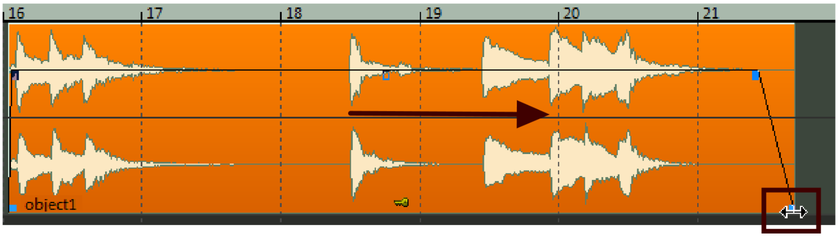
Now click the folder symbol in the object editor to the right of the name of audio file. You can select a new audio file from the dialog that follows.

Moving Audio Material Below the Object

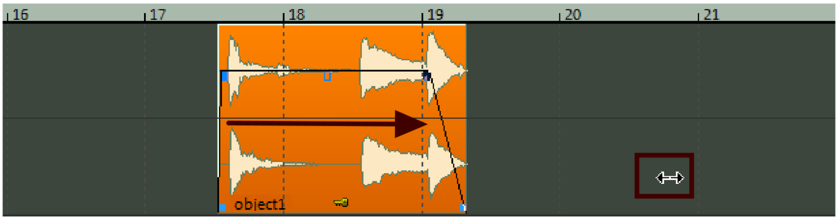
By pressing the right "Ctrl" button and clicking the lower part of the object, you can move the audio material below an object by clicking it. To do this, excess material must be available within the defined object borders, i. e. the audio file must be longer than the object that it is connected to.



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



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



External editing

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

You can configure this data excchange together with the corresponding context menu option with the help of the the dialogue "External Tools (view page 641)" which is located in the menu "File" > Program Settings" > "External Tools".

Manager

The individual managers are divided into functional categories and are used to bundle commonly used management and control commands in the respective manager windows.

The following manager windows are available:

Manager	Keyboard Shortcuts
File Manager	Ctrl + Shift + B
Object Manager	Ctrl + Shift + O
Track Manager	Ctrl + Shift + S
Marker Manager	Ctrl + Shift + Alt + M
Range Manager	Ctrl + Shift + Alt + B
Take manager	Ctrl + Shift + Alt + T
VSTi Manager	Ctrl + Shift + I
Routing manager	Ctrl + Shift + Alt + R
Soundpool Manager	
Infomanager	

To open a manager go to "View" > "Manager" and select the manager you want. The selected manager will open either in its own window or in the Docker (view page 90) if it is open.

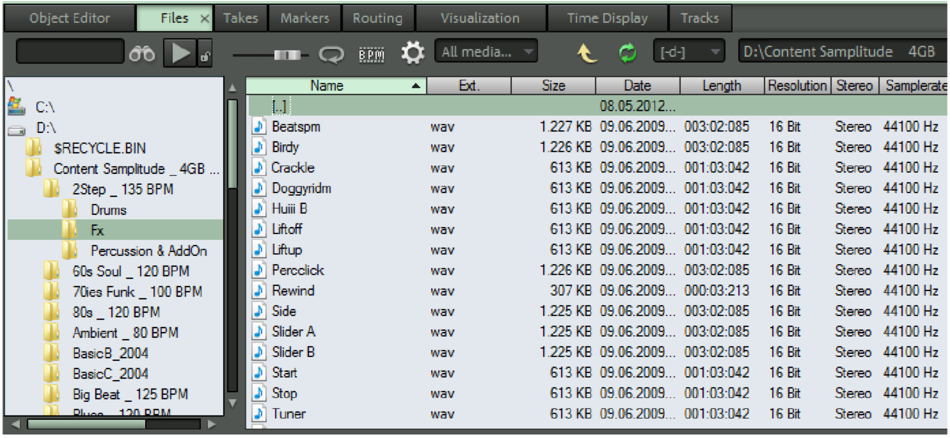
File Manager

Samplitude contains a file manager for previewing and loading files directly into projects. The browser can be used to create folders for favorites and provides access to all recently opened folders.

To display it go to "View" > "Manager" and select "File Manager". The manager will open either in its own window or in the Docker (view page 90).

Menu: View -> Manager -> File Manager

Keyboard Shortcut: Ctrl + Shift + B



Navigation/favorites

The file manager includes a directory tree that may be switched on/off; this behaves just like Windows Explorer. Directories and files are displayed with directory and file symbols in the form of a list.

The following file information is displayed:

- Name
- Extension
- Size
- Date
- Length
- Resolution in bits
- Stereo/mono
- Sample rate
- Timestamp
- Description/title
- Originator/artist
- Path
- BPM: Specified by the loop length

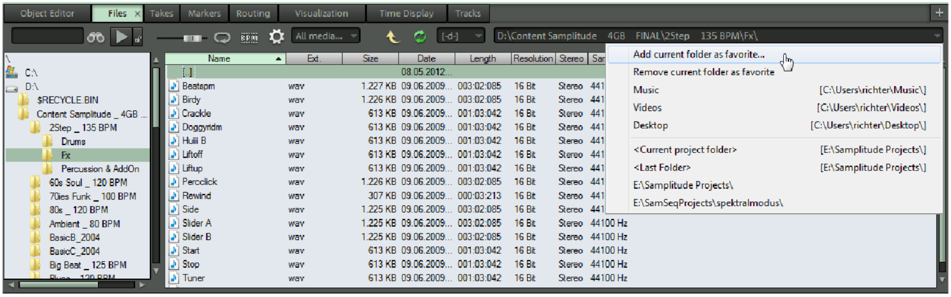
Navigation is done via mouse or keyboard. A drive selection menu located in the toolbar lets you quickly switch between all data storage devices connected to the workstation.

A display filter located in the toolbar limits the display to certain media types, e.g. .WAV, MIDI or project files, only. "All media files" is displayed by default.

Just like other manager windows, the "Search" option searches in the current window for directories or files. Enter a search item into the box and press the search button.

Any located directories or files are highlighted. You can find and highlight several entries that meet the search criteria by entering one or several first letters.

To the right in the file manager, the current directory may be added or deleted as a favorite to your favorites list. Click a favorite to open it.



Underneath the favorites list, a selection menu is located featuring a list of paths used last. You can switch into the current project folder here.

Preview audio files

A selected audio file can be previewed with the playback button in the file manager. Any highlighted audio file is played immediately if the “Autoplay" box is checked. A complete list can be previewed simply by using the cursor keys. Clicking the play button again stops continuous playback. The fader symbol can be used to control the volume.

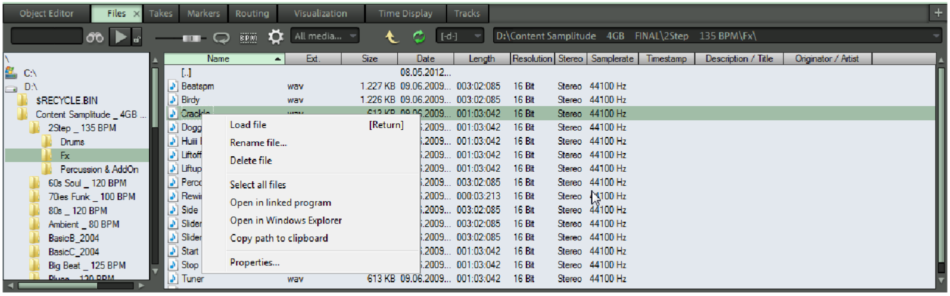
Playback does not proceed via the mixer or internal effects, but rather via the global playback device selected in the playback parameters (keyboard shortcut: "P"). If no playback occurs, check to ensure that the routing settings for the device are correct.

BPM sync: This option enables the selected loop to be previewed in the VIP's tempo. The original BPM is specified in this case by the length of the file. When playback is running, loops are positioned at the beginning of the beat by double clicking and then added into the arrangement as loop objects.

Looped preview enables the selected file to be previewed in a loop.

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

Loading Files into Projects from the File Manager



Media files in the file manager can be added to an open project by drag & drop or by double-clicking on the desired playback marker position. If the file manager list is in the foreground, pressing “Enter” produces the same result. Right-clicking the desired file opens a context menu. Here you can load, rename and delete files and folders.

Note: Deleted files are always sent to the Windows Recycle Bin. The keyboard shortcut "Shift + Del" plus a confirmation deletes the file permanently without sending it to the Recycle Bin.

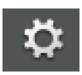
This also provides access to the functions "**Select all files**", "**Open in linked program**", **"Open in Windows Explorer"**, and "**Copy path to clipboard**". This enables you to display the respective file's **properties**.

Insert file into new track: The keyboard shortcut "**Alt + Enter**" creates a new track for the file. The track that is created is named according to the file.

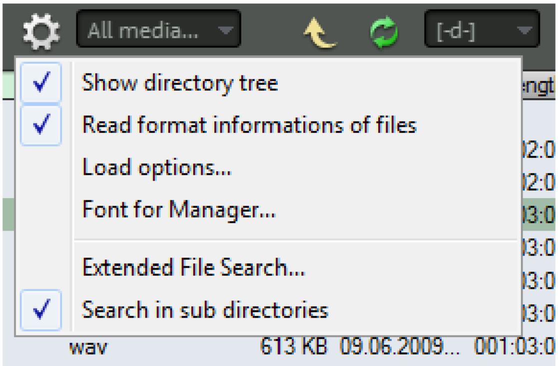
Open file in wave project view: Drag an audio file from the list and drop it into an empty space or into the title bar of the program window.

If several files are selected from the list, all files will be inserted at the current playback marker position as new objects with the loading options ("Options") taken into account. This way you can specify whether all files should be inserted into the current project one after the other, one underneath the other, alphabetically, or according to their timestamp positions. The load option settings are also valid for files loaded using the "Load" dialog.

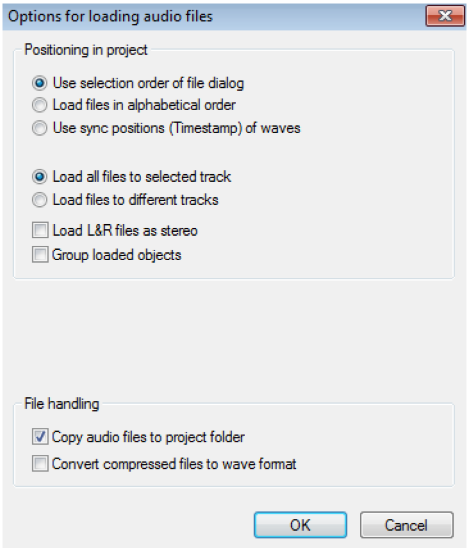
File manager options



Additional options can be accessed by clicking the small gear icon:



Loading Options



Use selection order of file dialog: If this option is activated, Samplitude remembers the sequence in which the files were selected and then sorts them accordingly.

Load files in alphabetical order: If this option is activated, Samplitude sorts the selected files alphabetically in the VIP.

Use sync positions (Timestamp) of waves: Broadcast wave files containing timestamps are positioned precisely at this position in the VIP.

Load all files to selected track: The selected files are loaded successively in one of the selected tracks.

Load files to different tracks: The files are now sorted in vertical order from the selected track to the next one. If necessary, an additional track is added.

Load L & R files as stereo: If you choose this option, Samplitude will load all L & R files as stereo files.

Group loaded objects: All loaded files are grouped. They can be ungrouped at any time.

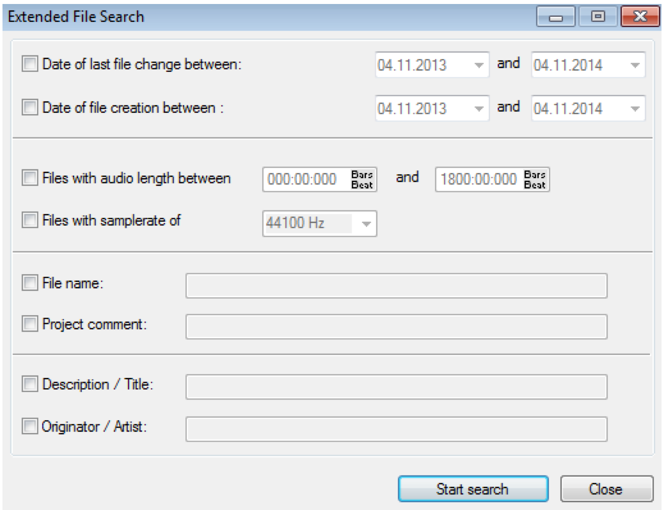
File Handling

Copy audio files to project folder: Copies the file to the corresponding project folder.

Convert compressed files to wave format: Compressed audio formats such as MP3 can also be loaded and played directly in Samplitude. This requires increased CPU calculation. Activate this option to convert files like this into an uncompressed wave format.

Extended File Search

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



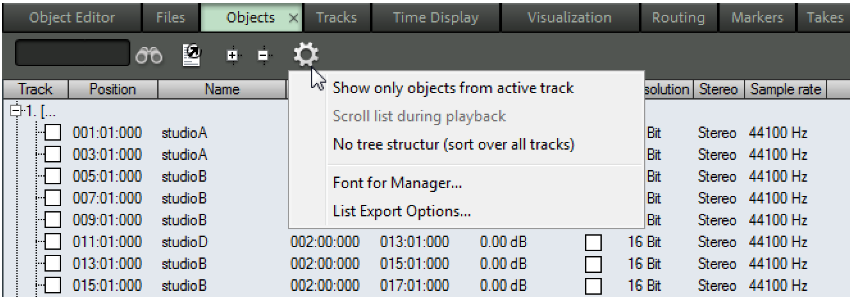
Object Manager

The object manager lists all objects contained in the current project and makes it possible to edit selected object parameters.

To display the object manager go to "View" > "Manager" and select "Object Manager". The manager will open either in its own window or in the Docker (view page 90).

Menu: View -> Manager -> Object Manager

Keyboard Shortcut: Ctrl + Shift + O



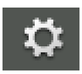
You can export object manager information as a text file. Click "Export text" in the toolbar to do this. The Windows text editor opens with an excerpt from the object manager list. The following information will be saved:

- Project name and project path
- Track and object name
- Start position in the project
- Path of source file

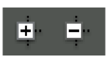
You can find this file in the project folder (Projectname.txt).

Object View and Selection

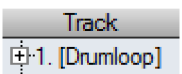
Objects are displayed in a tree structure according to tracks. Tracks are only listed if they also actually include objects. Further instances of the listed objects can be moved into the arranger using drag & drop. If you hold down the "Ctrl" key while doing this, a copy of the object will be created.



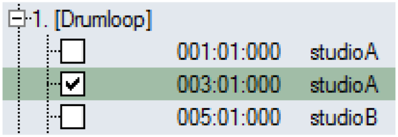
You also have the option of displaying objects in the current tracks only (toolbar: "Options").



You can use the +/- keys on the toolbar to open and close the track object display for the complete window.

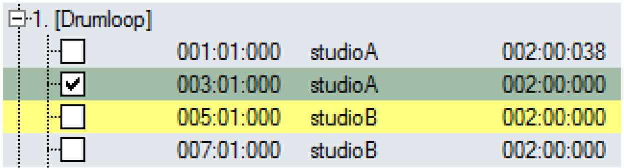


Click on the “+” symbol left of the track number to expand the view of a single track.



Select objects: The check box in front of the object name can be used to select one or multiple objects. This is also visible immediately in the project window.

If an object that is part of a group is selected, the other objects of the selection will also be selected in the project window and object manager. Grouped objects are also recognizable by the number of the object group in the "Group" column.



Objects that are currently playing are highlighted with color in the object manager.

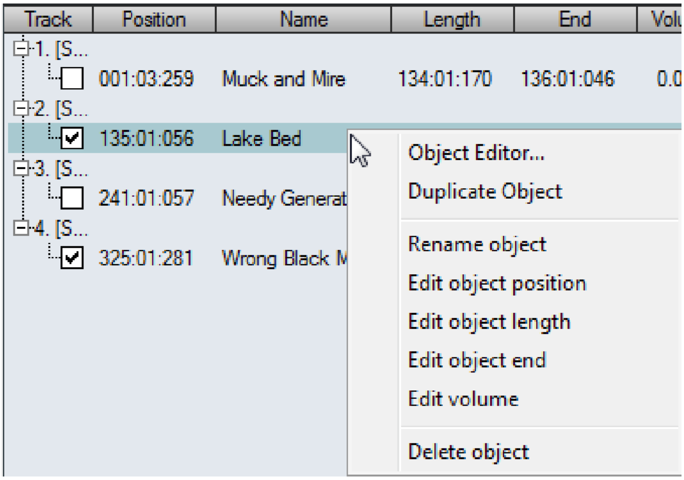
Search for objects: As in other manager windows, the Object Manager also provides a search option that you can use to search for objects in the current window. Enter a search term into the box and press “Enter”. Found objects are highlighted, but are not selected in the project window.

Delete, Rename and Edit Objects

You can delete objects directly in the object manager by selecting one or several objects simultaneously and pressing the “Del” key. Alternatively, you can delete an object in its context menu.

You can rename objects by double-clicking on the object names and entering a new name. You can also use the context menu.

To edit an object directly in the object manager, right-click the object entry you want to edit and start the object editor using the context menu that appears. In the same context menu you can also duplicate objects.



Edit object parameters

You can edit the following parameters in the object manager:

- Start position
- Name
- Length
- End
- Volume
- Group
- Lock

To edit a parameter, double click the respective entry and enter a different value. Numeric values can be changed by dragging with the mouse, and "Ctrl" and "Shift" allow larger or smaller changes. "Tab" moves to the next editable value. "Playback marker up/down" navigates vertically within a column, provided this is a text column.

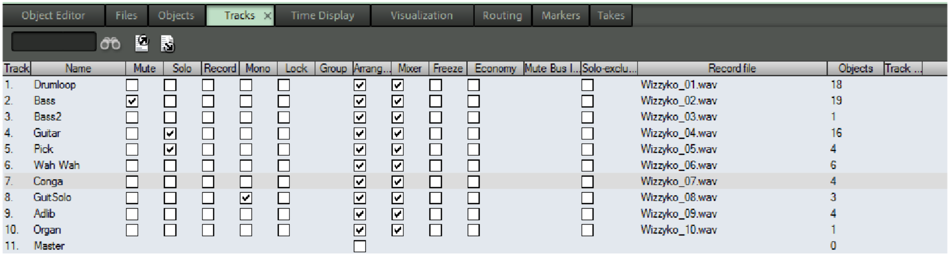
Track Manager

All the tracks in the current project are displayed in the track manager, which allows direct access to the "Solo", "Mute", and "Record" functions, as well as hiding tracks in the project window and mixer.

To display the track manager go to "View" > "Manager" and select "Track Manager". The manager will open either in its own window or in the Docker (view page 90).

Menu: View -> Manager -> Track Manager

Keyboard Shortcut: Ctrl + Shift + S



Track View and Selection

The track manager contains all of the tracks in the current project. To select a track, double-click on the track number. The selected track will be highlighted in the project window. If the selected track is located outside the visible area, the project window will automatically scroll to display it.

Search for tracks: As in other manager windows, the track manager also provides a search option that allows you to search for tracks in the current window. Enter a search term into the box and press “Enter”. Found tracks are highlighted.

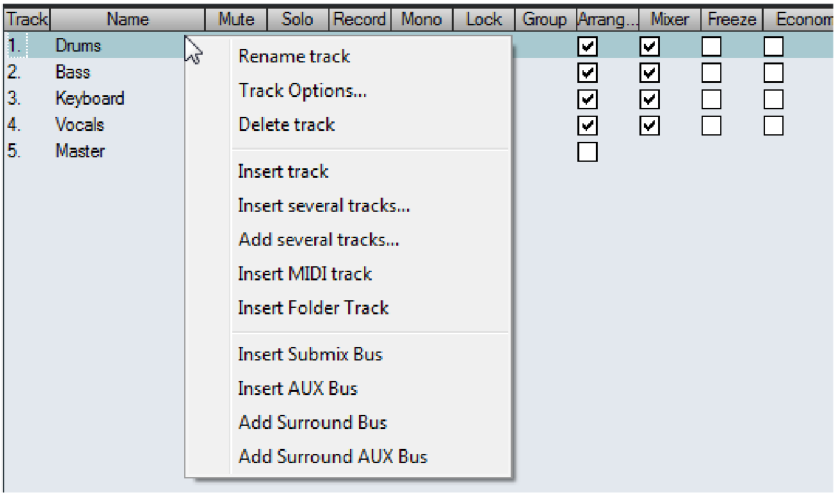
Sorting, Renaming, Deleting and Inserting Tracks

To sort tracks in the track manager, simply drag & drop them vertically to the desired position.

You can delete tracks in the manager by selecting one or several of these tracks and pressing the "Del" key. Alternatively, you can also use the context menu to delete a track.

You can rename tracks by double-clicking on the track names and entering a new name. You can also use the context menu for this. Pressing "Tab" moves the cursor to the next editable value. You can use the up and down arrows to navigate vertically within a text column.

New tracks can be added directly in the track manager. To do this, right-click on a track entry and select “Insert track” from the context menu. You can also add or insert folder tracks, submix buses, AUX buses, or surround buses. The "Track Options" are also made available by right-clicking.



Hiding and Displaying Tracks

In addition to direct access to the functions „Solo“, „Mute“ and „Record“, you can also hide tracks. To remove a track from the project window or mixer, click on the box in the “Arranger” or “Mixer” column to remove the check. Hidden tracks will still be played back and grouped objects on these tracks will still be edited.

To display a hidden track, reactivate the check box.

Write a Text File

Click the "Export text" button and Samplitude will create a text file with a listing of the tracks used in your project. The button beside it can be clicked to import a track list from a text file.

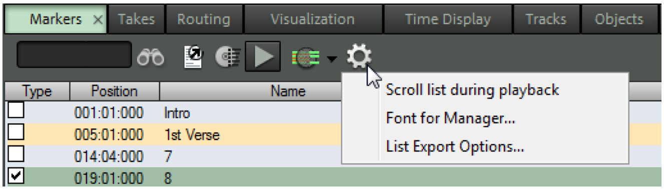
Marker Manager

The marker manager lists all of the markers contained in the current project and makes it possible to jump directly from the list to them or to play them.

To display the marker manager go to "View" > "Manager" and select "Marker Manager". The manager will open either in its own window or in the Docker (view page 90).

Menu: View -> Manager -> Marker Manager

Keyboard Shortcut: Ctrl + Shift + Alt + M



The following marker types can be displayed in the marker manager:

- Markers with numbers or names
- CD track markers
- CD subindex markers
- CD pause markers
- Source markers
- In/out point markers
- Tempo markers
- Beat markers
- AQ marker (audio quantization marker/transient marker)

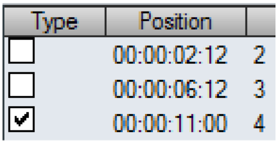
Marker manager information can be exported as a text file. To do this click on the "Export text" button in the toolbar. The Windows text editor opens with an excerpt from the marker manager list. The following information will be saved:

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

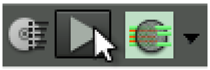
You can find this file in the project folder (Projectname.txt).

Playing from and jumping to markers

To position the playback marker on a set marker, click on the type column of the marker in the marker manager.



Play back from a marker: Select the marker and click on the "Play marker in a loop" button.



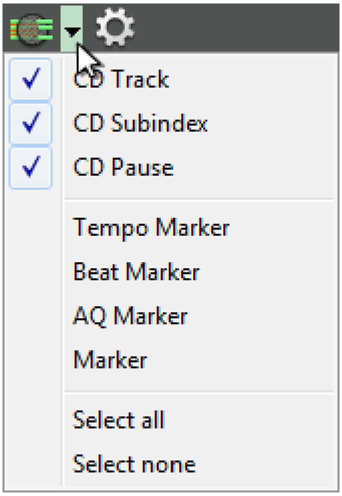
A range of two bars surrounding the selected marker is created and loop playback is started.

Marker View and Selection

Marker/CD Index Manager: This dialog lists all of the set markers and CD track indices in the project.

Detailed information about the Marker/CD Index Manager can be found in "CD/DVD" > "CD Track/Index Manager" (view page 1004).

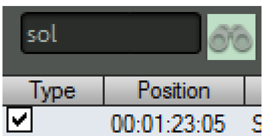
Marker filter: An optional marker filter lets you limit the view to selected marker types. In the filter section click on all of the marker types you want to display. All marker types that are not selected will be hidden.



Export text: If you click on the „Export Text“ button, Samplitude will create a text file with a list of all the markers in the project.

Search for markers: As in other manager windows, the marker manager also provides a search option which allows you to search for markers in the current window. Enter a search term into the box and click on the binoculars button.

Found markers will be displayed.



Right-clicking on the marker list opens a context menu where you can specify the CD track, CD subindex, CD pause, CD end or position markers. In the context menu you can also insert new markers, rename markers, adjust marker positions and delete markers.

Set range between selected markers: In the context menu you can set a range between the selected markers.

Deleting markers, changing a marker's name, position, and type

You can delete markers directly in the manager by selecting one or several of these markers and pressing the "Del" key.

You can rename markers by double-clicking on the marker names and entering a new name, or you can use the context menu.

To change a marker position double-click on "Position" before entering a new position.

The "Tab" button switches to the next editable value. "Arrow up/down" navigates vertically within a column.

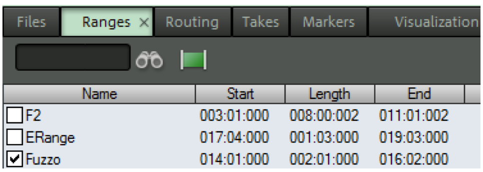
> **Tip**: You can change the type in the context menu for the respective marker any time, e.g. if you want to change a position marker into a CD track marker.

Range Manager

The range manager displays all of the ranges in the project and makes it possible to select them directly from a list.

To display it go to "View" > "Manager" and select "Range Manager". The manager will open either in its own window or in the Docker (view page 90).

Menu:	View -> Manager -> Range Manager
Keyboard Shortcut:	Ctrl + Shift + Alt + B



Define and Search for Ranges

Defining Ranges: To save a range in the manager, you first have to define a range in the project window (view page 130). Next, click "Define new range" in the toolbar of the range manager. Ranges that are saved using the program's “Save Range” feature (Alt+F2, Alt+F3, etc.) appear in the list appended with F2, F3, etc.

Search for Ranges: As in other manager windows, the range manager also features a search option that lets you search for ranges in the current window. Enter a search term into the box and click on the binoculars button or press Enter. Found Ranges are highlighted.

Edit Range Parameters

You can edit the following parameters in the range manager:

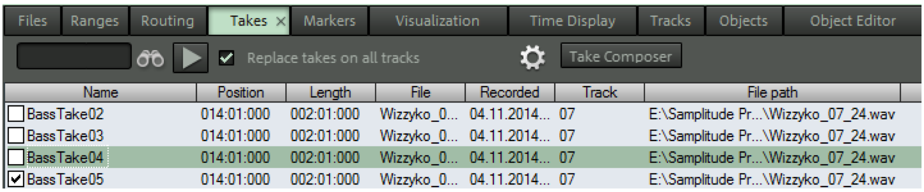
- Name
- Start
- Length
- End

To edit a parameter, double-click the value and enter a different value. Numeric values can be changed by clicking and dragging; "Ctrl + Shift" enables larger or smaller value changes. Pressing "Tab" moves the cursor to the next editable value. You can use the up and down arrows to navigate vertically within a name text column.

Take manager

Menu: Tools - > Manager -> Take manager

Keyboard shortcut: Ctrl + Shift + Alt + T



The take manager offers you a convenient option for selecting and organizing recording takes, loop recordings (view page 49), and destructive object edits (view page 164).

When graphical comping with the take composer (view page 192), you can edit together a new take from available take sections.

> **Note**: A new take is also added after effects calculation and this is displayed immediately in the take manager if the option "Create copy" is selected in the destructive effects dialog.

Take manager – basic working principle

In addition to the audio and MIDI data, Samplitude also saves additional information like the time position and the track where the data was recorded in each recorded file.

This data is stored as take information in the wave project and the MIDI object. For multitrack recordings, the takes also contain information like which other tracks were involved in the recording.

This enables each recorded passage to be assigned to a section, marking it as a take of a certain recording.

First, record as many takes of a certain passage as required. If a new take in an arranger track is recorded over an old one at the same position, no existing data will be overwritten. The new recordings will be attached to the end or saved to a new file.

If an object from a recording is selected and the take manager is opened, takes from the same track and original position are searched for in the whole audio material of the project, and the results are shown as takes.

Note: The take manager only accepts objects that were created by recording, but not those created by wave or CD import.

Take Manager – Application Examples

- To select the best takes after some punch loop recording sequences
- To find the best takes from several recording sequences of a production by means of predefined bar positions.
- To display a clear overview of all available takes in a specific SMPTE period.
- To edit the best parts of existing takes of recorded vocals into a "perfect take"

The basis for working with the take manager is always a selected object, e. g. the most recently created object after a punch-in recording. The current object take can be identified by the checkmark preceding the names in the take list. Now all the takes suited to the selected object are shown in the list. In the basic settings, this refers to all the takes from the same track and original position.

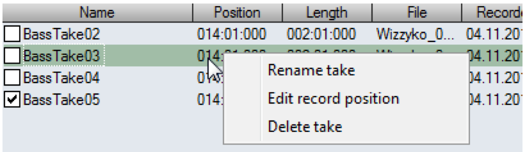
Select Takes

To select a take for the object, place a check mark before the desired take, or press Enter after you have selected the corresponding take in the take manager.

Select a take directly in the arranger window by applying "Ctrl + right click" to the object.

All of the takes shown in the take manager and the currently selected take are arranged one below the other in the take composer (view page 192). Takes can be compared directly in the take composer and a combination of the various takes can be edited together.

Rename take: Right-click on the take to open the context menu and rename it or edit the recording position in the take manager. Multiple selection is also possible.



Edit recording position: This option enables you to make changes in the position field of the take manager.

Delete takes: Right-click on the take to open the context menu and delete it. Multiple selection is also possible. Once you have deleted all of the takes except for one, you can decide if you would also like to delete the referenced file. Note that the deleted audio file will not be usable in other projects.

Note: If the audio file contains several takes but is not referenced through an object, it can be completely deleted from the hard drive without having to delete each take. To do this, use the "Remove unused samples (view page 594)" function in the "Clean up" section of the File menu.

For a better overview, easy-to-use take-naming options are available in the recording window (view page 732) even during recording.

Take manager – options

Filter recording position: Only takes that overlap the same original time positions are shown, e.g. the selected object.

Filter out takes that are too short: Only takes that are at least as long as the currently selected take are shown.

Show all tracks: Takes of all tracks will be displayed.

Manager font: Use the dialog to specify the font, format, and size for the take manager

Take management for multitrack recordings

For multitrack recordings, each recording creates associated classified takes on each track. All simultaneously recorded objects are grouped and also given the same take name.

Switch takes on all tracks: This allows all takes of a multitrack recording to be exchanged in one step.

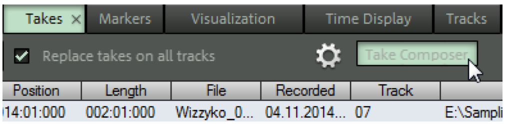
Note: Since the take manager display refers to the last clicked object, the reference track may be switched at any time.

Take Composer

The Take Composer is a function of the Take Manager that displays a selected track's individual object recording runs (takes) and the created Revolver Tracks (view page 142) below each other. Produced Revolver Tracks are located below the takes.

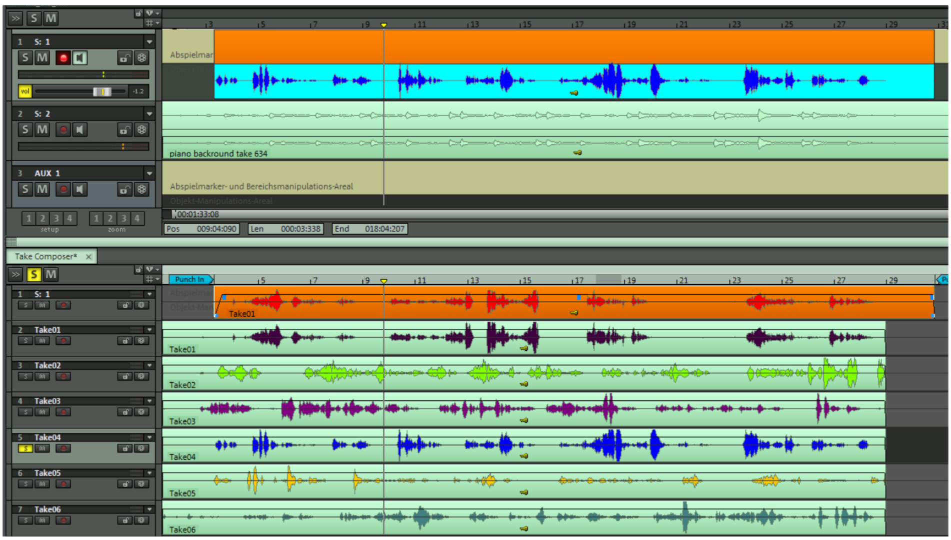
Use the Take Composer to add the best parts of individual recordings of an object to create the perfect take, or edit the Revolver Tracks for the corresponding track.

Open the Take Composer by clicking the corresponding button in the Take Manager.



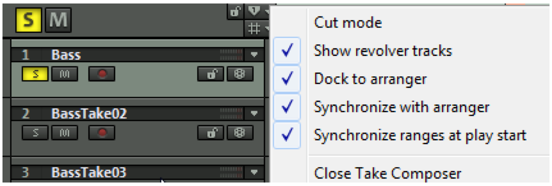
Note: Please make sure to select a take object before opening the Take Composer.

By default the Take Composer opens as an extra window below the arranger.



"Punch-in" and "Punch-out" markers will be displayed for the last selected object in the arranger.

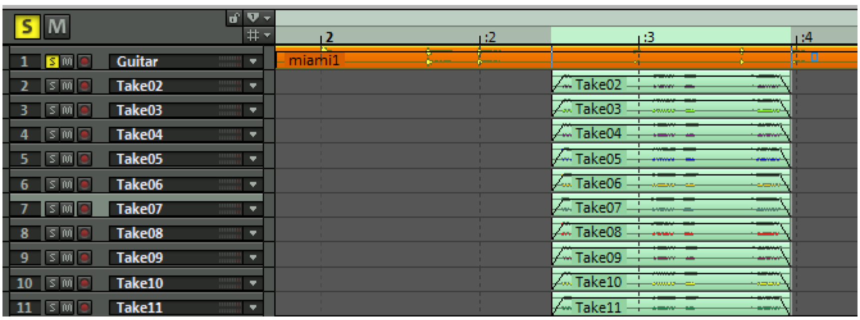
In the Take Composer use the space bar to initiate playback in sync with the arranger.



In the Take Composer's marker menu you can specify whether you want to display revolver tracks (view page 142), whether the Take Composer should open in its own window or should be docked to the arranger, whether the Take Composer should run in sync with the arranger and whether the last selected range in the Take Composer should be synchronized with the arranger before playback.

Here you can also switch to cut mode to quickly execute "comping" (view page 194) (see below) or close the Take Composer.

The current take selection for the respective track is displayed in the first track of the Take Composer. During "comping", you can edit the best passages of the individual takes together in this "Comping Track" by using the editing operations described below.



The individual recording takes are arranged in the Take Composer 2 onwards under each other.

Monitoring with the arrangement

To listen to the recorded passages individually, switch the corresponding track to "Solo" and start playback. The solo mode "solo exclusive" is used automatically. Start the playback with the spacebar, the take is played together with the arrangement.

Monitoring without the arrangement

To play a take without the arrangement, press Ctrl + Spacebar to start playback.

Note: If you select an object in the Take Composer, you can listen to only that object with "Ctrl + Space", ignoring the "Solo" settings for the take tracks.

If multiple takes should be played back simultaneously in the Take Composer, click the corresponding track header and press the shortcut "Alt + S". This switches the track to (non-exclusive) solo mode.

Comping

"Comping" refers to the process of combining several takes to create a single one.

Comping in cut mode

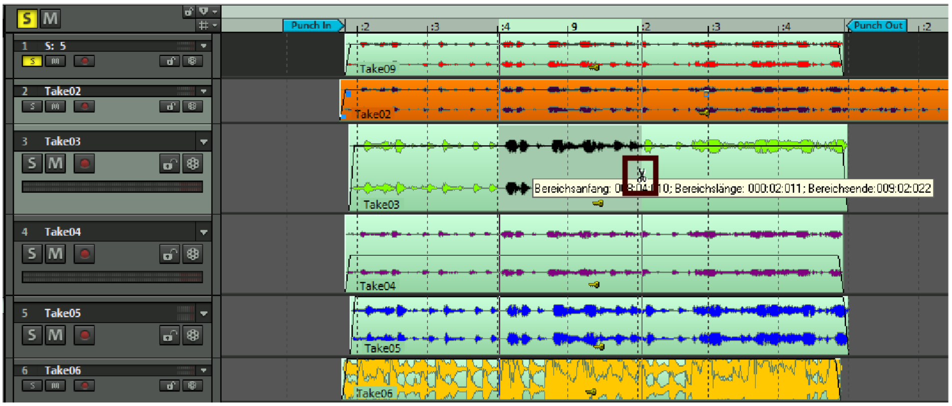
Switch to the mouse mode Cut mode (view page 110), which works a little differently than usual in Take Composer for comping. The mouse pointer becomes the scissors icon.

Now click on the desired cut point in the respective take. The selected take is inserted in the first track of the take composer from the current position up to the end. Auto crossfade (view page 663) mode is activated as a default during the process. Cut the desired sections from the respective takes in this way into the top track along the timeline.

If you only want to transfer specific ranges of a take into the comping track, drag a range over the respective take with the scissors tool. When the mouse button is released, the selected range in the respective take will be copied to the first track.

To replace an object of the "Comping track", select a different take with Shift+Click below the object to be replaced.

Tip: This way you can also change the first object of the comping track that comes from the active take when you open the Take Composer.



To play a take from the cursor position in "Cut" mode, use the keyboard shortcut "Ctrl + mouse click".

Comping in "Universal" mode or "Range" mode

You can also comp in Universal Mode or Range Mode: Drag a range over a take and transfer it to the first track using the keyboard shortcut "Shift + C".

If no range is selected, the shortcut "Shift + C" will insert the selected take from the current position up to the end.

When the take composer project is closed, changes can be confirmed, which transfers the take manager's comping track into the original project.

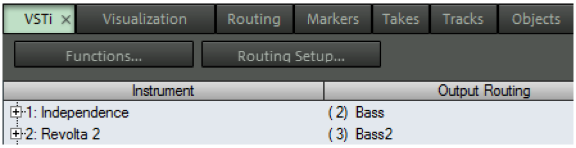
VST Instruments Manager

The VSTi manager enables easy display, insertion and deletion of VST, MAGIX, and Rewire instruments in the current project.

Menu:	View > Manager > VST Instruments Manager
Keyboard shortcuts:	Ctrl + Shift + I

Right-clicking on the selected instrument opens the plug-in dialog of the selected VSTi. If the plus sign beside the instrument display is clicked, the outputs to the assigned tracks will be shown. By right-clicking the "Output Routing" column you can select a track that will only receive the output of the respective individual outputs. The assigned output then appears in the plug-in slot of the target track.

The general instrument outputs are routed in the VSTi manager in the "Routing Setup". You can specify how the outputs will be routed in the dialog that appears. You can either route all outputs to the current track or create new mono or stereo tracks for each output. Pressing the "Del" key removes a selected instrument completely from the project. If only individual output signals are selected, routing to a track can be canceled by pressing the "Del" key.



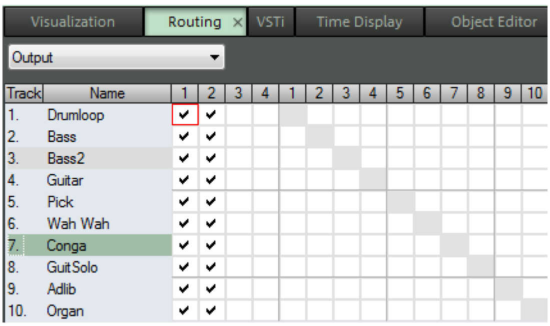
Detailed information about instrument routing settings (view page 414) and about managing single outputs is provided in the chapter "Routing VST instruments with the VSTi Manager".

Through "Functions" you can access the "Freeze" and "Unfreeze" commands of the integrated VST instruments. Here you can "Create new tracks for single VST outputs" as well as "Reset or delete selected instruments or outputs".

Routing Manager

The routing manager provides you with a clear matrix display of the inputs, outputs and AUX sends and VCA groups of all tracks.

Menu	View > Manager > VST Instruments Manager
Keyboard shortcut:	Ctrl + Alt + Shift + R



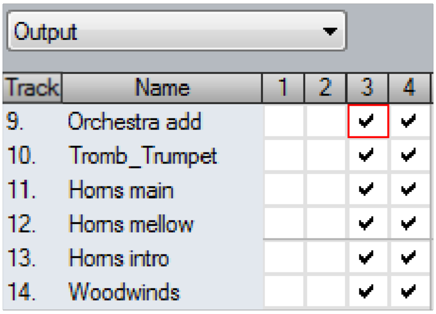
By simply clicking in the matrix fields you can assign all available inputs and outputs to your project's tracks or lift existing routings.

By right-clicking you can define the selected output as a "Direct out" (red fader display), a "Pre-fader out" (yellow fader display) or "Post-fader out" (orange fader display) or open the stereo panorama dialog to set the panorama (view page 229) for the respective outputs.

This is how you can assign multiple sequential tracks to the same inputs/output (vertical):

Step 1: Select the input/output of the first assigned track by clicking on the corresponding Matrix field (e.g. "track 9", "outputs 3+4").

Step 2: Assign the same input/output using Shift + mouse click to the Matrix field of the last track to be assigned, e.g. "track 14", "outputs 3+4").


Output

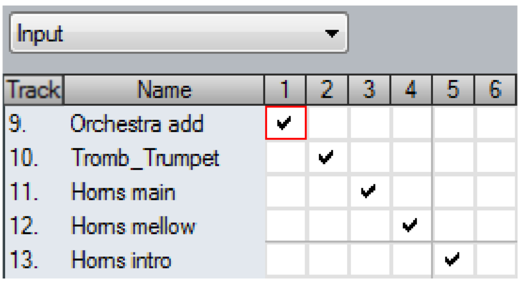
Track	Name	1	2	3	4
9.	Orchestra add			✓	✓
10.	Tromb_Trumpet			✓	✓
11.	Horns main			✓	✓
12.	Horns mellow			✓	✓
13.	Horns intro			✓	✓
14.	Woodwinds			✓	✓

In vertical routing all selected and intermediate tracks are assigned from top to bottom to the same input or output.

This is how you can assign multiple sequential tracks to the next row of inputs/outputs (diagonally):

Step 1: Select the input/output of the first track to be assigned by clicking the corresponding matrix field (e.g. "track 9", "input 1").

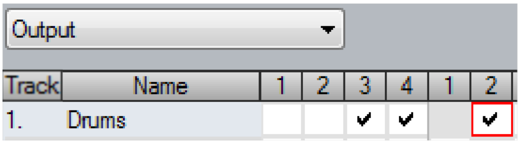
Step 2: Following the diagonals in the matrix, assign the corresponding input/output using Shift + mouse click to the matrix field of the last track to be assigned (e.g. "track 13", "input 5").


Input

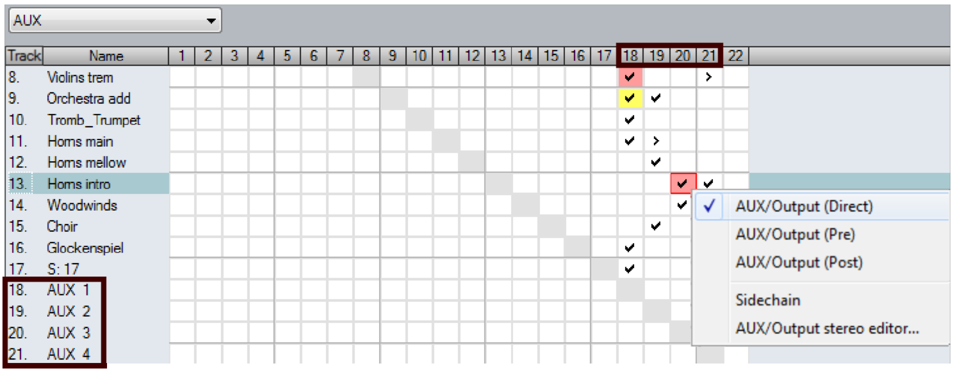
Track	Name	1	2	3	4	5	6
9.	Orchestra add	✓					
10.	Tromb_Trumpet		✓				
11.	Horns main			✓			
12.	Horns mellow				✓		
13.	Horns intro					✓	

In diagonal routing, all selected and intermediate tracks will be assigned to corresponding inputs and outputs in sequential numbering.

In the output section, you can repurpose any track as a submix bus. Clicking on the corresponding matrix field behind the available output transforms the track into a submix bus.

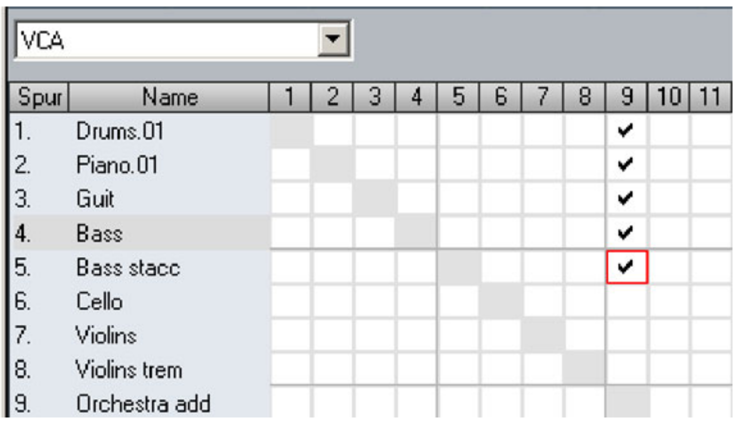


You can also route Aux busses to the individual tracks (view page 226). By right-clicking you can define the selected AUX bus as a "direct out" (red field), "pre-fader send" (yellow field) or "sidechain send" (view page 228) (">" symbol), or open the stereo panorama dialog to set the panorama (view page 229) for the respective AUX bus.



VCA Group Assignment in the Routing Manager

Here you can select each track as VCA Group (view page 214) and assign it as many tracks as you like.

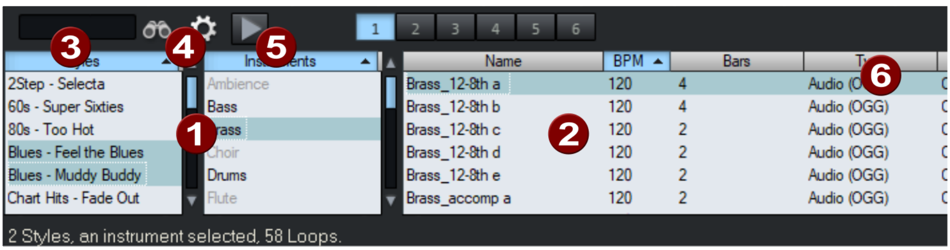


In the example above, you define track 9 to VCA group. By clicking on the "VCA" button in the respective mixer channel, you can see that the status of the fader has changed: "Fader is VCA master". Tracks 1 to 5 are controlled by the VCA group in track 9.

Soundpool Manager

In the Soundpool Manager you can integrate you Soundpool loop content in a clear form. 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).



1 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. Above this, the different pitches are displayed (if available).The list of found samples results from the selection of the entries 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.

2 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.

3 **Full text search:** Above right in the search field you can search the list of samples found for a specific sound file name.

> Tip: If you want to search in multiple lists parallel, open a new manager via "Tools -> Open new manager..." and switch to Soundpool in the new window. A separate search query can be set for each list. A separate search query can be set for each list.

4 You can access a selection menu by clicking on the cog symbol. This menu provides several additional options for maintenance and display of your Soundpool database:

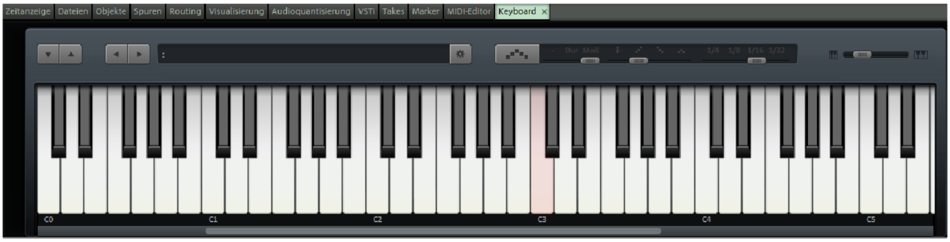
- **Add styles:** This option allows you to add new MAGIX sound pools. To do this, select the folder or disk where the sound pools are stored.
- Normally, loops can be previewed just by clicking them, even during project playback. If **Automatic playback** is deactivated, then use the playback button in the manager. 5.
- 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.

6 There is one principle difference between the types of loops: While audio loops (.wav or .ogg) are normal (looped) audio objects that can be applied anywhere in the arranger, MIDI loops (MIDI takes) consist of 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 loop in another pitch is possible.

Keyboard

The keyboard enables software synthesizers to be played and recorded directly via the on-screen keyboard.

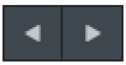
Menu:	View > Keyboard
Keyboard shortcut:	Ctrl + Alt + Shift + K



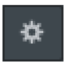
You can click the keyboard with the mouse to play the instrument. The closer to the bottom edge you click on the "virtual keys", the louder the sound will play. Of course, you can't seriously play music by clicking with the mouse (this function is more suitable for testing out sounds quickly). That's why you can also play the keyboard with the keys on your computer's own keyboard.

Note: This works only after you first click once on the keyboard using your mouse. Otherwise, pressing the computer keys will function as keyboard shortcuts for the different functions in Samplitude. 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 so that the keyboard may be played using the computer keys.



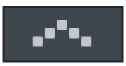
You can use the horizontal arrow keys to select the next/previous sound of the synthesizer, or they may be selected directly in the list field on the side.



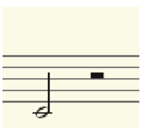
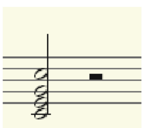
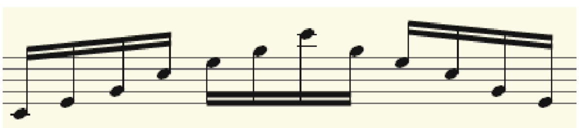
This button opens the editor window for the synthesizer for fine tuning the sound.

Use the slider to the right to drag the graphic display of the keyboard buttons wider. This is especially useful if you want to operate the keyboard using a touchscreen.

Arpeggiator



The arpeggiator is a special function that can be used to create full chords or broken chords (arpeggios, i.e. the notes of a chord played in quick succession) by pressing a single key.

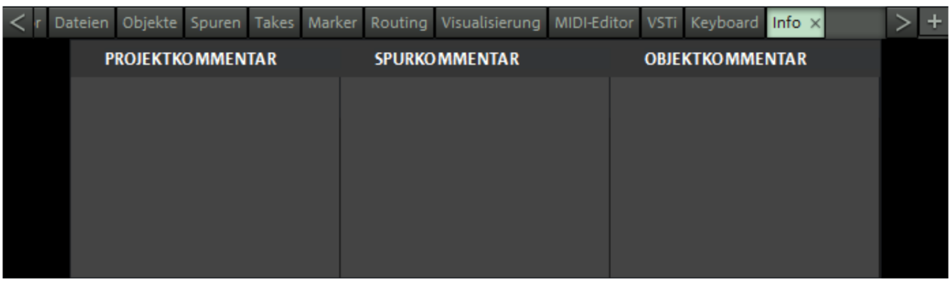




C note *C minor chord* *C minor arpeggio with 1/16 notes*

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. When set to the far right, a normal chord is played. The other positions are up, down or up and down. The figures are repeated as long as the note is played.
4. The tempo of the arpeggio is set here and can range from 1/4 notes (slow) up to 32nd notes (very fast).

Info manager

The info manager displays notes for individual objects, tracks or the entire project on a clearly-arranged interface.



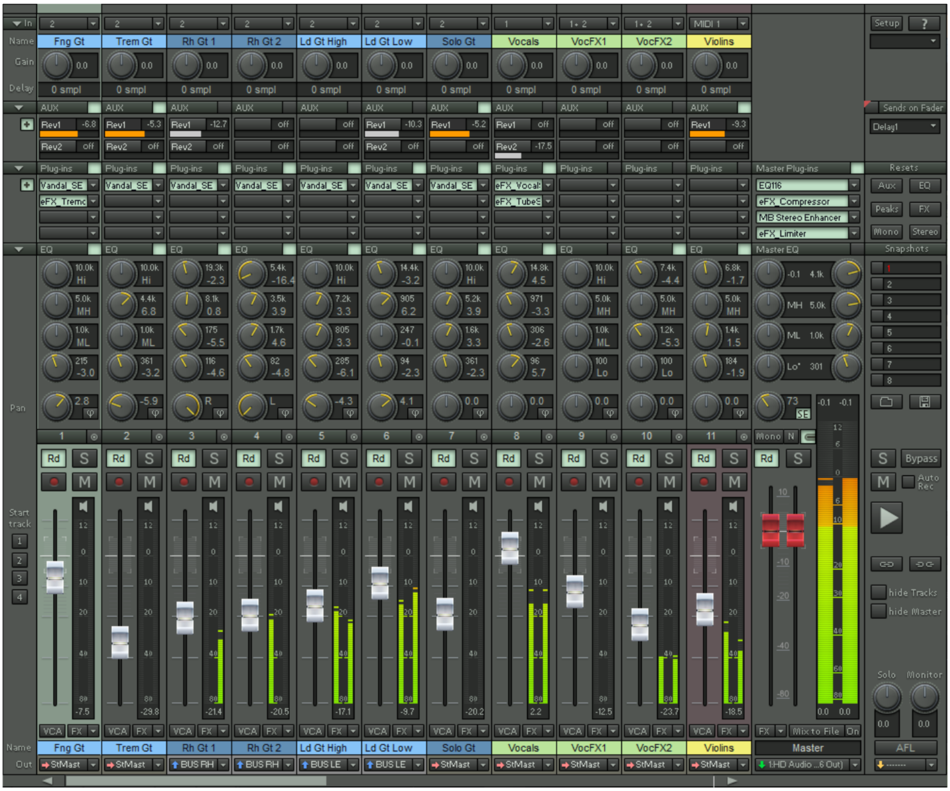
You can write notes concerning the currently selected project, track or object in the three fields.

Mixer

This function has been revised in Samplitude. For the latest information, see the PDF document **Samplitude Pro X7 New Features** in the program folder.

You can open the Samplitude mixer by going to "Window > Mixer"(shortcut: "M"). This is largely equivalent to a hardware mixing console. The integration of the Samplitude mixer into the digital environment is still much more flexible than an analog mixer.

All settings that you make in the mixer are calculated and edited in realtime. This includes track and master effects, panorama and volume settings, initial assignments for single tracks/busses as well as initial assignments for the master section.



Using the mixer

Every VIP track has a corresponding mixer track. However, you can hide any VIP or mixer track in the track manager (view page 182).

Alternative Mixer Skins

In the System Options (shortcut: „Y“) under „Design > Skins (view page 613)“ you can choose from various interface designs, a.k.a. "skins". A further option for changing the mixer skins is to click on the Samplitude icon in the top left corner of the mixer window.

Mixer - keyboard shortcuts

- **Arrow keys:** The arrow keys on your keyboard can be used to navigate through the individual mixer elements and to select the active mixer element (fader, knob or switch).
- **Page up/Page down:** Changes the value of the active mixer element. Hold the "Ctrl" button at the same time to change the value in large increments and hold down the Shift key for smaller increments.
- **Home:** Sets the mixer element to its preset value. If you press "Home" again, the element is reset to the last value, providing a simple comparison between the edited and preset states.
- **End:** Opens the assigned sub dialog of an element, e. g. the EQ window for one of the EQ knobs. This function corresponds to clicking on a knob with the right mouse button.
- **Enter:** Opens the numeric entry field for a controller. Switches are turned using the "Enter" key and "Page up/Page down" keys.
- **Del:** An activated plug-in slot can be reset with the "Del" key.

Mixer Operation with Mouse and Keyboard

Left-click: Selects a control element.

> **Hint**: To move a single control with the mouse wheel you dont have to select it first. Just mouse-over the knob or fader and turn the wheel to change its value! for finer changes additionally press the Shift key.

Right-click: Accesses either a context menu for the corresponding control element or a dialog with additional settings options.

Ctrl + left-click: You can use this shortcut to select multiple control elements (multi-selection).

Shift + left-click: You can select all similar control elements between the previously selected and the current control element (multi-selection).

Note: After selecting several mixer elements, you can form these into one group (view page 219). Use the "Group selected controls" button to do this.

Ctrl + Shift + left-click on knobs and faders: This keyboard shortcut reverses the behavior of corresponding faders or knobs within a linked group (inverse selection).

For example, this feature is useful for moving two grouped faders in opposite directions with a single mouse movement, or for controlling the opposing movement of the panorama controllers on two tracks simultaneously.

Alt + Left-click: A single control element can be temporarily positioned within a group by pressing the “Alt” key and operating the element.

Double-click on numbers: Opens the data entry field to allow manual entry of numeric values.

Double-click on knobs: Resets the control element to its preset (initial) value. Double-clicking again sets the value back to the altered value again.

Clicking on value range: Clicking on the area below the faders or on the left of knobs decreases the values. If the mouse button is held down, the value is automatically reduced until the button is again released or the end of the scale has been reached. (Keyboard shortcut: Page down).

Clicking on the area above the faders or on the right of the buttons increases the value. If the mouse button is held down, the value is automatically increased until the button is again released or the end of the scale has been reached. (Keyboard shortcut: Page up).

The level faders can be adjusted very precisely. If you click on them while keeping the left mouse button pressed and moving the mouse pointer left or right from the fader and moving the mouse in a vertical plane, you can fine tune the settings. The step size of the settings become smaller as the mouse pointer is moved further away from the selected fader.

The step size of the fader movement also becomes smaller if you hold down the "Shift" key while moving a fader.

Knobs (potentiometers) can be set in two different ways. After clicking on a knob you can move the mouse around the knob or move the mouse up and down as for the faders. This makes it possible to reduce the step size value by holding down the Shift key. The way in which the knobs are controlled can be set in the "System Options" dialog (keyboard shortcut: Y) under "Keyboard, Menu & Mouse > Mouse".

Drag & drop in the mixer: You can rearrange individual channel strips in the track number field or name field using drag & drop. The mouse pointer turns into a hand during this operation. The overall settings of a channel can be transferred to another channel by dragging & dropping the "FX" button. The equalizer settings can be transferred by dragging & dropping the "EQ" button.

The plug-in slots offer drag & drop functionality to copy effects to a different track or between tracks and the master. The sequence of effects for VST and DX plug-ins can be changed within the track using drag & drop.

Note: If a certain effect is already available, a repeated drag and drop of the same effect onto the same track performs a reset of the effect to the values transferred with the effect and does not create an additional copy of the effect; the original values are overwritten by this process.

If all you want to do is move effects between the channels in the mixer instead of copying them, simply keep the Shift key pressed while dragging & dropping.

Channels

Each track in the VIP is assigned to a specific mixer channel. The sound settings of the respective mixer channel are applied to the objects on the track.

Every channel strip offers the following settings options whereby the individual sections can be opened and closed by clicking on the corresponding arrow symbol:

Input: Shows the audio input for the recording (see below).

Gain: Controls input amplification on the mixer for the channel.

Aux Sends/Outputs/Sidechains: In this section the AUX components for various AUX buses (view page 225) can be activated and set. Open a menu by right clicking for advanced settings such as AUX pan editor (view page 226). You can also switch to output display (view page 227) or sidechain send display in this context menu. The button at the top right turns all signal paths either on or off, which effectively makes it an bypass switch.

A newly added AUX bus is always displayed in the arranger at the bottom and in the mixer to the right with the highest channel number.

Plug-ins: Here you can insert effects into the channel. Clicking on the empty insert slot opens the plug-in browser (view page 240), where you can load a plug-in to the slot. Click on an occupied slot to activate/deactivate the plug-in. Right-clicking on the slot opens the interface of the plug-in. The arrow next to the respective insert slot opens a menu with various functions. You can ope the plug-in browser again to switch out the plug-in or remove it ("No effect"). The "Plug-ins" button activates or deactivates all effects in the channel. A visual indicator (*) is displayed for plug-ins that were previously active and are active the next time "Plug-ins" button is pressed.

Equalizer: Here you can adjust the frequency response of the track using the EQ 116, a fully parametric 6-band equalizer.

You can numerically edit the gain and frequency of the corresponding filters.

For fine-tuning, right-click one of the knobs. Samplitude shows the equalizer dialog for you to specify more exact settings.

Panorama: This adjusts the signal alignment in the stereo panorama. The settings of the knob have a different effect on mono and stereo tracks. In a mono track, depending on the control setting you can place the signal more left or right in the stereo panorama. In contrast, in stereo tracks, this lets you control the balance between the left and right channel. If you are working with a surround track, the position in the surround field can be set here.

Beside the panorama dial, you'll also find a switch to reverse the phase.

Right-clicking on this opens the stereo panorama dialog (view page 229) where you can make adjustments to the panning laws and the stereo width.

Link: This button connects the corresponding channel with the one to the right of it. All faders, panorama, input, AUX sends, as well as EQ settings and adjustments now affect both channels. Both panorama faders are linked through inverse selection. With the keyboard shortcut Ctrl + Shift + left-click on knobs you can invert the behavior of corresponding panorama knobs within a linked group.

Automation: This button gives you access to the channel's automation functions

> Detailed information about automation is available in the chapter "Automation (view page 428)".

Solo: The solo button mutes all channels with the exception of the selected channels.

> Keyboard shortcut: Alt + S

More specific Solo button behavior can be set by right-clicking on it:

Solo exclusive: This setting switches the active track to "Solo exclusive" mode, i.e. only the active track is audible. All other tracks are muted in "Solo" mode.

> Keyboard shortcut: Shift + Alt + S

Solo safe: This setting ensures that all channels that are switched to solo are automatically monitored with the AUX Return channels that they feed.

> Detailed information on the global solo modes can be found in "Global buttons -> Global solo modes (view page 224)".

The output assignment for the channel also appears in the same context menu.

Recording: This button activates the track. Right-click to specify the recording mode for the input signal (see above "**In (Input)**").

Mute: The mute button mutes the selected channel.

> Shortcut key: Alt + M

More specific Mute button behavior can be set by right-clicking on it:

Mute/Inactive: Mutes and also deactivates the selected track. This increases performance as caching and FX processing of this channel are not necessary.

Keyboard shortcut:	Ctrl + Alt + M

Mute bus inputs: Use this function to mute the bus inputs routed to this channel.

The output assignment for the channel also appears in the same context menu.

Volume fader: Controls the track levels. The behavior of the MIDI track's volume fader is preset to controller 7 (MIDI volume). Alternatively, you can also specify that volume fader setting does not change for MIDI tracks or that the volume fader for MIDI tracks should match the MIDI velocity scaling (view page 326). To do this, right-click the track's volume fader.

Note that a virtual synthesizer's audio output can usually be created, edited, and mixed on the same track as the MIDI data the instrument is receiving. In some cases, this results in duplicate use of the volume fader which, on the one hand, controls velocity or MIDI volume (CC 7), and on the other, controls the audio level. These are not identical parameters. For instance, a MIDI instrument played with high velocity can be added to the mix at low volume and vice versa. For this reason, the volume fader can be assigned differently.

Note: Levels higher than 12 dB may also be entered numerically. To do this, double-click on the volume value display at the bottom end of the channel fader scale and type in the desired value.

Monitoring: Click the loudspeaker symbol to activate the monitoring function. This will route sound card input signal through to the output.

If the "Hybrid Engine" is set up for monitoring, all mixer channel effects for the output can be used. This requires ASIO drivers for the sound card. By this, Samplitude can be used as a live effect device, for example.

More details on Monitoring can be found in the chapter "System settings >Monitoring settings (view page 72).

VCA: This button activates the VCA Mode for the channel. Here you can set the fader as the VCA master or assign the fader to a new VCA group ("VCA Groups (view page 214)").

FX: This button activates the Effect Routing dialog (view page 239) for the corresponding track and can be used to configure and edit the effects chains. This determines the sequence of the effects.

More information about this is available in the chapter "Buses and routing -> Order of effects calculations and signal manipulation".

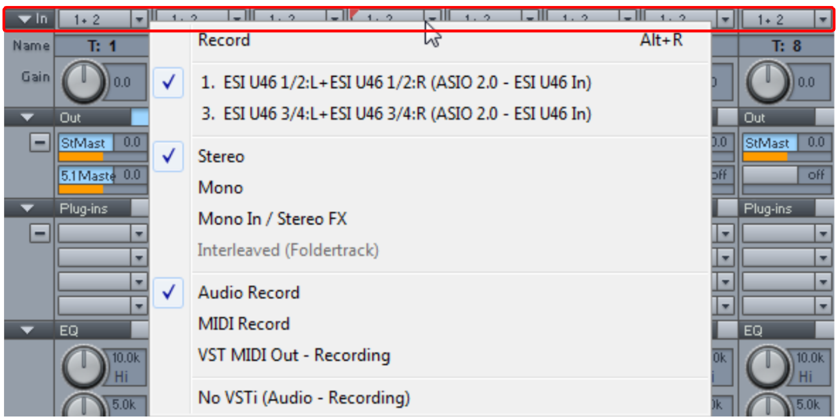
Right clicking the "**FX**" button opens a context menu. Here you can open the effects routing dialog for the channel and the preset effects settings. These can be copied, inserted, reset, saved, or loaded. You can save your own track effect settings in the folder „Program Files" > „MAGIX" > „Samplitude" > „fx-preset" > „Track FX". You can also create new subfolders here.

Track name: You can enter the name of the VIP track here. This name can be edited by double-clicking.

Out: Sets the output for the channel. This can be a stereo master channel, another stereo audio output, a submix bus and/or a surround bus. For MIDI channels, you can set the MIDI (VST) output device here. The option "No Output" means that the channel is not assigned to any output.

In (Input)

Determine the recording mode for the incoming signal up top on the channel strip.



- The "Record" option activates the track.
- Below you will see all the available inputs for your audio/MIDI interfaces. Select the input where the signal you want is located.
- If you select "Stereo In", the channel input will be switched to stereo and the signal will be recorded on two channels.

- If you select "Mono In", the channel input will be switched to mono and the signal will only be recorded on one channel.
- If you choose "Mono In / Stereo FX", the channel input is switched to mono and a single-channel input signal is recorded, but the track itself remains a stereo track in which the effect calculation is performed in stereo.
- You can select whether you want to record in audio or in MIDI and - if using MIDI - whether you need a VST MIDI output.
- In addition, if you have selected "Audio recording" above, the program will display "No VSTi (audio recording)".

Apply effect to multimple channels simultaneously.

You can apply multiple instances of the same effect to multiple mixer channels:

Step 1: Select the channels that you would like to apply the effect to. First click on the channel number of the first desired channels to activate it. Then, holding the Shift key, select the number of the last channel from the row. This will select all channels in between. If you press the Ctrl key instead of Shift, you can add any additional channel to the selection.

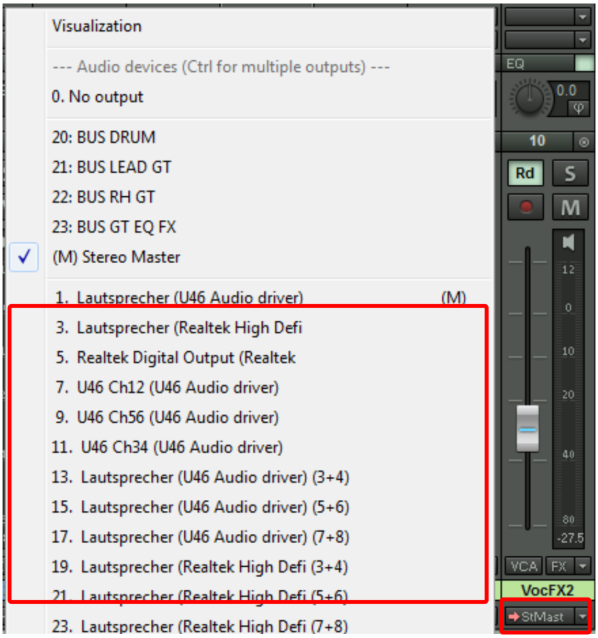
Step 2: Now, select an insert effect for one of the selected channels and make your adjustments. As soon as you close the effect dialog window, the effect will be integrated into the insert slots of all selected channels with the exact settings made. Additional parameter changes will be applied to all instances of the effect found in channels selected at that time.

Step 3: To remove the effect from all selected channels, simply take it out of one insert slot. All other instances of the effect will also disappear.

Assign track to multiple outputs

You can also route the output signal of a channel to several hardware outputs, buses or masters simultaneously, for example to create a stereo mix in the same project in addition to a surround mix, or to listen to a mix on different speaker systems.

Click on the corresponding field in the output section, hold the Ctrl key and select additional audio outputs.



The stereo master output is labeled with and (M) and is the default output.

If you set the AUX section of the channel to "show outputs" with a right click, you will get a quick overview of the outputs used.

To revoke assignment to a stereo output, click to the corresponding field in the output section, hold the "Ctrl" key and select the stereo audio output you would like to deactivate.To revoke assignment to a stereo output, click to the corresponding field in the output section, hold the "Ctrl" key and select the stereo audio output you would like to deactivate.

In the Matrix display of the Routing Manager (view page 196) you'll find a good overview of all assigned outputs.

VCA Groups

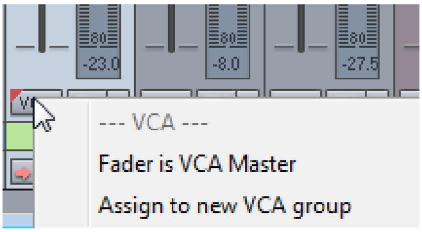
VCA is the abbreviation for "Voltage Controlled Amplifier". In analog mixers VCA are found as voltage controlled amplifiers. This makes the fader influence the music signal indirectly, i.e. through a decoupled voltage stabilizer. VCA groups are a type of a remote control that makes it possible to control many individual channels and mix them together. No audio channel flows through VCA groups themselves.

A major advantage of VCA groups is that, the way the VCA Master Fader regulates both the assigned channel signals and all post fader AUX send feeds are down regulated too. In this way the balance between the direct and effect signal remains the same.

In contrast to audio groups, VCA groups have their own audio output, meaning they have audio outputs at their disposal, and offer the possibility to further edit the sum total signal with insert effects.

You can also work with VCA groups in Samplitude. You can be access the functions using the VCA buttons of the individual channels:

By clicking on the VCA button in the channels a context menu will appear which shows you which VCA Master channel (VCA group) controls the fader.

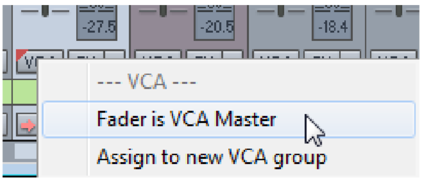


Create a VCA group

Step 1: Create a new channel in the mixer by right clicking on the AUX channel number and selecting "Insert tracks" > "Insert empty track".

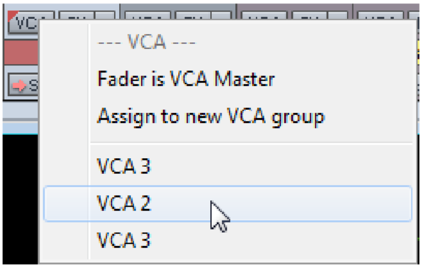
Step 2: Double click on the name field of the new channel and name it "VCA1".

Step 3: To properly turn this channel into a VCA group, press the "VCA" button and select "Fader is VCA Master".



Assigning channels to an available VCA group

Click on the VCA button of the channel that you want to control using a VCA group and select the VCA group which should control the channel.



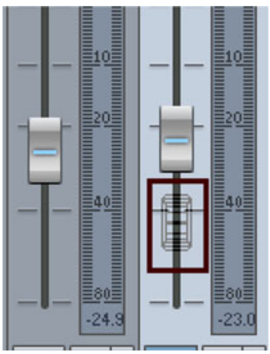
Assigning channels to a new VCA group

Here creation of a VCA group and channel assignment to a VCA group are united in one option. If you want to integrate a specific channel into a new VCA group, select the VCA option "Assign to new VCA group" for the corresponding channel in the mixer. At the back end of the mixer a new VCA group channel is created, to which the selected channel is assigned.

Ghost Fader

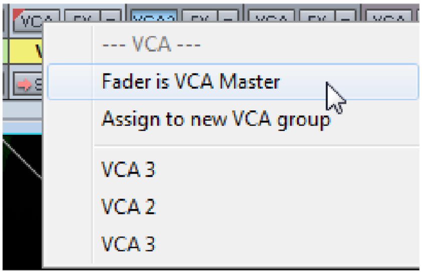
By assigning individual channels to a VCA group "Ghost Faders" are created which display the resulting volume in the track according to the set VCA Master Controller. This way the VCA master control also influences the volume of assigned channels. Relative volume levels of channels in relation to each other remain unchanged.

This enables continued control of the volume of every individual channel assigned to a VCA group. The ghost fader value will change accordingly.



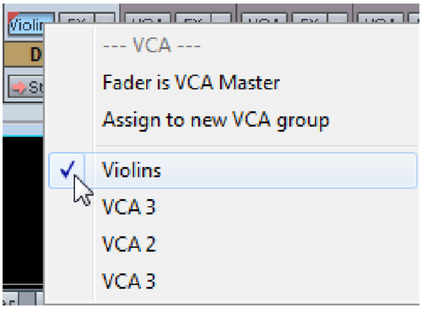
Change channel fader into VCA Master

Each fader can serve as VCA master for other channels. As soon as you select the "Fader is VCA Master" option for a channel, this channel will appear as a VCA Group in the dialog selection.



Remove VCA mapping

Remove a channel's VCA assignment to a VCA Group by switching off the selected VCA group in the respective channel. To do this remove the corresponding tick in the VCA button's context menu.



When deactivating the "Fader is VCA Master" option in a VCA Master all assignments for this VCA group will be revoked.

Assigning and deleting VCA assignments can be done in the Routing Manager (view page 198).

Master Section

The master section can be found to the right of the channel strips.

Master plug-ins: This function can be used to set effects inserts, MAGIX plug-ins, and VST plug-ins for the master output signal. Right-click on the button to open the plug-in dialog (view page 239). To switch the plug-in effects on or off quickly, click on the button.

Master Equalizer: This equalizer affects the complete signal.

Stereo Enhancer (SE): The multiband stereo enhancer is used to modify the entire signal's stereo image. Three bands are available for the bass, mids, and high frequencies.

Mono: This button plays the entire signal in mono. This can be used to text mono compatibility.

Normalize (N): This is the master normalization. Clicking this button adjusts the output level so that the loudest parts of the signal reach 0 dB. The maximum level reached during the last playback process will serve as the basis during this process and will be displayed on the peak meter.

Note: If the level display is clicked in the stopped state, the playback marker will jump to the position indicated by the level.

Automation: This button gives you access to the channel's automation functions.

Detailed information about automation is available in the chapter "Automation (view page 428)".

Link: This button connects the left and right channels of the master signal.

Faders: These faders adjust the left and right master signal. Double-clicking on one of the faders sets them both to the 0 dB position.

FX: Opens the effects/routing dialog for editing and configuring any number of effects chains. This determines the sequence of the effects.

More information about this is available in the chapter "Buses and routing -> Order of effects calculations and signal manipulation".

Mix to file while recording

The Mix to file and On buttons can be used to mix down in realtime and to change any of the mix's parameters during playback. At the end of the playback process, the mixdown created will be written to a wave file.

1. Click the "Mix to file" button to open a dialog to specify the name and place where the created wave file should be saved. When the "On" button is active, the master output of the mixer can be written to a wave file during playback.
2. Now start playback of the VIP
3. On playback, any parameter may be changed in order to record any live adjustments that are made.
4. At the end, stop the playback of the VIP.

Important: Make sure that the "Mix to file – on" button is deactivated if the master signal should not be mixed into a new file. Otherwise, the system creates a wave file every time the file is played.

The "Mix to file" function does not necessarily need to be used to record dynamic mixer movements during playback. Effects may be controlled via AUX busses, and volume and panorama curves may be automated.

Note: The "Mixdown" button is available in some mixer skins, e.g. "Multitrack mixer" or "Recording mixer". This function also mixes down and saves the entire VIP, including all settings, to a single file. In contrast to "Mix to file", no parameter changes during playback are taken into consideration, but the track is merely "bounced". Detailed information may be found in the menu reference under "Menu tools -> Track bouncing (internal mixdown)".

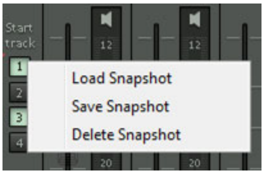
Device/master out: Enter the playback device for the stereo signal. Select "Master inactive" if the master playback device should not be used in the project (e.g. multiple I/O setups).

Note: If the settings for the master playback device are changed, all tracks routed to the master will also automatically change settings. If this automatic process is not helpful, select "Master inactive" before setting up a new device.

Global Buttons

Some additional buttons for adjusting the global settings of the mixer window and mixer handling can be found to the right side of the mixer window.

This includes (counter-clockwise from bottom left):



Start track: If the arrangement has many tracks, only a section of the corresponding mixer channels are generally displayed in the mixer window. The visible section can be moved using the scroll bar at the bottom of the mixer.

Hold the "Shift" key down and click on one of the "Start track" buttons to save the current view (a left click is sufficient the first time). The previously saved section can be recalled by clicking the "Start track" button again. Right click the "Start track" button to open a context menu to select between "Load snapshot", "Save snapshot", and "Delete snapshot".

Solo/Monitor/PFL/AFL/Solo in place (view page 222)

View functions: Here you can hide or show the individual tracks and the master section.

Group selected controls: These two buttons are used to group and ungroup elements of the mixer.

To group controls, select the desired elements while holding down "Ctrl" or "Shift" and then activate "Group selected controls". To ungroup, select one of the elements in the group and click on "Ungroup controls".

Note: If you have previously selected multiple tracks, this function is also available to group selected controls if the same control elements are involved. More information about grouping tracks in Samplitude is available in "Working in the project window -> Selecting/grouping multiple tracks (view page 139)".

Multi-channel Selection: Multi-channel selection in the mixer functions exactly like multitrack selection in the arranger, i.e. by clicking the track number/name while holding down the "Ctrl" or "Shift" key. In order to clear a group of channels, click a channel before or after the selection and activate a control element.

Play/stop: Playback of the VIP can be started and stopped with this button. Right-clicking opens the transport console.

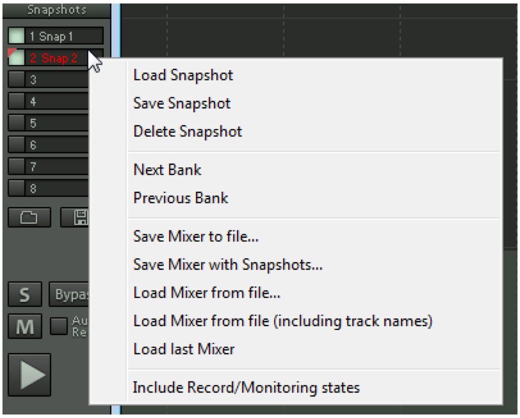
AutoRec: This display will light up if the level automation is written in the master.

Bypass: Using this button you can switch off active effects in the project for comparison purposes. You will now only hear the audio with the volume and panorama settings for the mixer channel.

Solo/Mute: All solo/mute functions can be activated and deactivated globally using these buttons.

Load mixer setup/save mixer setup: These buttons enable the current mixer settings to be saved as a preset (without snapshots) or previously saved mixer setups to be accessed.

Snapshots: The snapshot function is particularly useful for comparing various setups. You can easily switch between setups. Naming snapshots is possible by double clicking the name field. Right-clicking a snapshot opens a context menu to save, load, or delete snapshots or to switch snapshot banks. Using this process, up to 32 mixer snapshots can be saved.



Save mixer to file: Save mixer settings to a separate file using this option.

Save mixer with snapshots: This option enables mixer settings to be saved including snapshots. When these mixer settings are loaded in the future, a dialog will ask whether the saved snapshots should also be included. If this is confirmed with "Yes", then you will overwrite any snapshots used up until this point.

Load mixer from file/load mixer from file (including track names): Complete mixer settings can also be accessed with or without track names.

Load last mixer: When loading a snapshot, the current mixer settings will be saved temporarily and these can be recalled with the command "Load last mixer". This makes an A-B comparison between the snapshot and the settings possible.

Reset (mono): Resets all mixer settings to default settings for mono tracks (when using mono or LR audio files). This reset function refers to the track panorama settings (view page 229).

Reset (stereo): Resets all mixer settings to the default settings for stereo tracks (when working with stereo audio files). This reset function refers to the track panorama settings (view page 229).

Reset AUX: Resets all AUX portions of the channel strips to their default values (= no AUX portion).

Reset EQ: Resets all equalizer settings.

Reset peaks: This enables the LED peak meter to be reset (peak hold display).

Reset FX: This enables you to remove effects from the signal path for all channels.

Sends on Fader: With this function you place the small sliders for one AUX path at a time on the channel faders. This makes it easier to set and compare the aux send amount for the tracks. (This corresponds to the AUX page on digital mixers, as these usually do not have separate AUX controls)

Select an AUX channel from the list box to activate the function. The function can be switched on and off by clicking on the yellow button.

?: Click this button to open the help file, which includes detailed information about the mixer.

Solo / Monitor Volume Control

In the master section there is a solo level and a monitor level available.

The solo knob controls the monitor volume of the solo mode, while the monitor knob also influences the level on the monitor output. The value corresponds to the Volume setting in the Monitoring section (view page 234).

Furthermore, the audio monitor position can be set using the AFL/PFL switch.

The selected monitor output corresponds to the first output device of the monitoring section (view page 232).

The signal can be accessed at two different positions:

PFL: PFL (pre-fader listen) intercepts the signal before the faders and effects, but after the input gain.

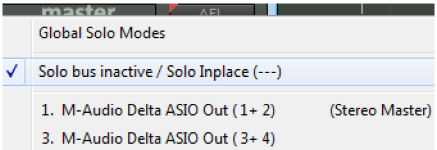
AFL: AFL (after-fader listen) takes all panorama changes/fader movements into account and influences the monitoring signal accordingly.

In general, the monitor bus behaves like "Main to Monitor", meaning that the content of the master can be heard at the same time on the monitor bus. As soon as solo is activated, only the solo state is played over the monitor bus.

Solo in Place (default)

This solo button behavior has been in use in Samplitude since the beginning. "Solo in place" sets channels to solo in the mixer at their corresponding positions within the stereo picture of the mix. Simultaneously, all other channels are muted. This solo mode is typically used during the mixdown to identify single instrument tracks and edit them individually.

You can switch to "Solo in place" mode and simultaneously deactivate the solo output bus by clicking on "Solo bus inactive/Solo in place" under the AFL/PFL button.



Attention: When using the monitor fader in this mode, it is possible to influence the audible master level and thereby the level of external devices connected to the stereo master. This means you can set up the monitor level in the mixer without using any external volume controllers. The audible result is different to that of the peak meter display in the mixer (which is the reason the numerical level values in the mixer are displayed in red). Additionally, the solo fader influences the volume of the solo tracks that are played through the master section. However, the internal level for bouncing and burning remains unchanged.

A further feature of this mode in Samplitude is that you can toggle between Pre-Fader Listen and After Fader Listen in Solo in place mode by clicking on the AFL/PFL switch.

Solo/monitor bus (in the "Hybrid Engine" only)

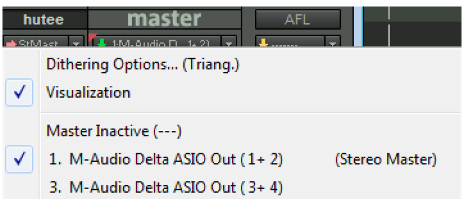
Use this bus to monitor tracks individually by pressing the corresponding solo buttons without influencing the stereo sum signal. This also affects mastering or live mixing.

The signal can be accessed at two different positions:

PFL: PFL (pre-fader listen) accesses the signal before the faders and effects, but after the input gain.

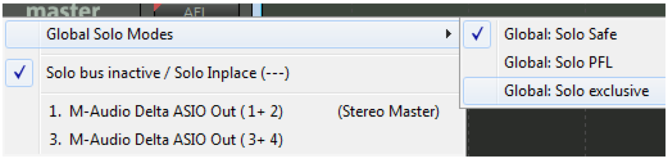
AFL: AFL (after-fader listen) includes all panorama changes/fader movements and influences the monitoring signal accordingly.

To set up an AFL/PFL monitoring line, switch the button to AFL or PFL by clicking it, and then select an output in the slot under your sound card as a solo bus.



Global solo modes

You can use the solo bus selection to set the global presets for using solo. This can also be accessed by right-clicking the solo buttons.



Global: Solo is not explicitly listed as a mode. If no check is set for a solo mode, then global solo will be active and you will hear the solo channels without AUX return channels.

Global: Solo safe: While in "Solo in place" mode, every channel switched to solo may be heard including the AUX return channels automatically.

Global: Solo PFL (available only in hybrid mode) switches solo tracks to PFL mode. The signal is accessed before the fader and effects, but after the input gain. When deactivated, the solo tracks are accessed after the fader.

Note: Economy tracks cannot be monitored with PFL.

In Global: Solo exclusive mode, only one channel may at a time may be switched to solo by clicking the solo button. In this case, the solo state is deactivated for all other tracks. If you activate the "solo" button for another channel, only this will be switched to solo. Solo is deactivated for the channel previously switched to solo. This enables you to quickly "walk through" individual channels.

To listen to the AUX return channels on the channels switched to "solo", select "Global: Solo safe" mode as well.

Note: While in "Global: Solo exclusive" mode, to still be able to listen a single channel in "Solo exclusive" mode, click the channel's "solo" button while holding down the "Shift" + "Alt" keys.

If you have already set "Solo exclusive" as the global solo mode, the same key combination deactivates "Solo exclusive" mode in order to hear multiple channels "solo" at the same time.

Buses and routing

AUX buses

An AUX bus combines all signals of the corresponding AUX sends from the individual channels. AUX buses are usually used for controlling real-time effects via the volume curve. For this purpose, part of the signal of the desired mixer channel is sent to the AUX bus ("AUX send") where effects are added. The fader of the AUX bus represents "AUX return".

Note: AUX tracks normally don't contain objects in the arranger view. They provide objects on other tracks with effects.

Submix buses

A submix bus combines several tracks. It controls the volume, panorama, and effect settings for the sum of all tracks that are "routed" to the submix bus. For example, all drum tracks (hi-hat track, bass drum track, etc.) may be combined as one submix bus so that the entire drum kit may be controlled via the volume controller of the submix bus.

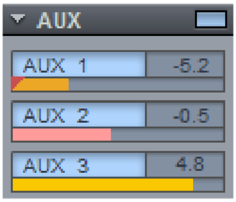
Creating submix/aux buses

Submix or AUX busses are always meant for stereo busses. The outputs may be routed for any channel to output devices or other submix buses.

- To create an AUX bus, click with the right mouse button on the number of the channel in the mixer. In the context menu, select "Insert tracks -> New AUX bus". Another option for adding an AUX bus is to raise the volume of an unused slot in the AUX area of the track editors or the mixers. This creates a new AUX bus automatically.
- If an AUX bus has already been created, you can the feed the signal of the previously created AUX section into the the newly created AUX bus.
- To create an Submix bus, click with the right mouse button on the number of the channel in the mixer. In the appearing context menu, select "Insert Tracks -> "New Submix Bus". The submix channel is integrated behind the channel that is selected. All channels, whose outputs are routed to this bus will be edited as a sum in this channel.
- Right clicking the number of a channel provides a context window for selecting between the properties "AUX bus", "Submix bus", or even both for the respective channel strip.

AUX Routing

In the AUX section of the Mixer and Track Editor you can route AUX sends and control their levels.



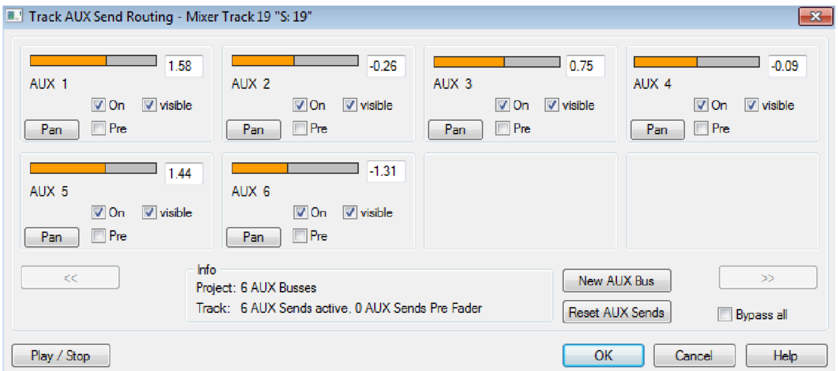
In the context menu of the Aux section you can set the AUX modes Direct out , Pre-Fader or Post-Fader.

Direct out: Audio is sent to AUX directly after gain. The color of the AUX send fader changes to red.

Pre-fader: Audio is sent to AUX after gain and all fx processing but beforethe volume fader. The color of the AUX send fader changes to yellow.

Post-fader: Audio is sent to AUX after all effects and the volume fader. This is the default setting. The color of the AUX send fader is orange.

Right-clicking on a mixer channel's AUX button opens the AUX routing dialog.



You can also open this dialog by right-clicking on the channel number and selecting "Track Effects > AUX Sends...". You can enter the intensity of each AUX bus in numeric values or by moving the orange bar (the bar is not visible in Off mode).

By default all AUX buses in Samplitude are set to "Post Fader". To switch them to "Pre-Fader", place a checkmark in the "Pre" box. You can use the effects routing dialog (view page 239) to specify the precise position of the "pre" and "post" AUX buses within the effects chain.

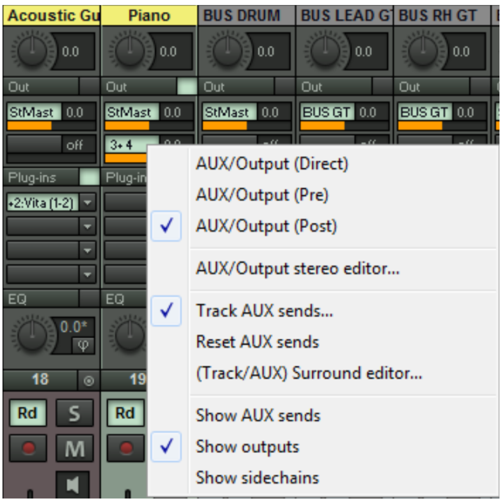
The "Pan" button applies a complete panorama section for every AUX send, which operates analogous to the stereo panorama dialog (view page 229) of a track. For example, this allows you to adjust the stereo width or invert the phase for the AUX send.

New AUX Bus: Creates a new AUX bus.

Reset AUX Sends: All AUX parameters are reset.

Displaying Outputs

In the AUX section of the Mixer and Track Editor you can also display the output assignments of the channels with a right click. This is especially helpful if you use multiple outputs (More information can be found in the section "Assign track to multiple outputs" on page 212).

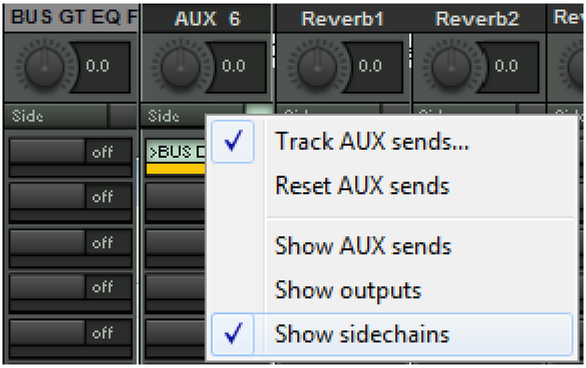


By right-clicking you can define the selected output as a "Direct out" (red fader display), a "Pre-fader out" (yellow fader display) or "Post-fader out" (orange fader display) or open the stereo panorama dialog to set the panorama (view page 229) for the respective outputs.

In the Matrix display of the Routing Manager (view page 196) you'll find a good overview of all assigned outputs.

Show sidechain sends

With a right-click in the AUX section of the mixer and Track Editor you can also display the output assignments of the sidechain sends.



In tracks that are used as sidechain signals or mixer channels, a new sidechain bus will be created and a sidechain send fader will be assigned for the target track. To differentiate from the "normal" AUX buses, a "**>**" is added. "Pre-fader" is set to catch the side chain signal (yellow fader display). This means that the sidechain remains independent of the fader position of the sending channel.

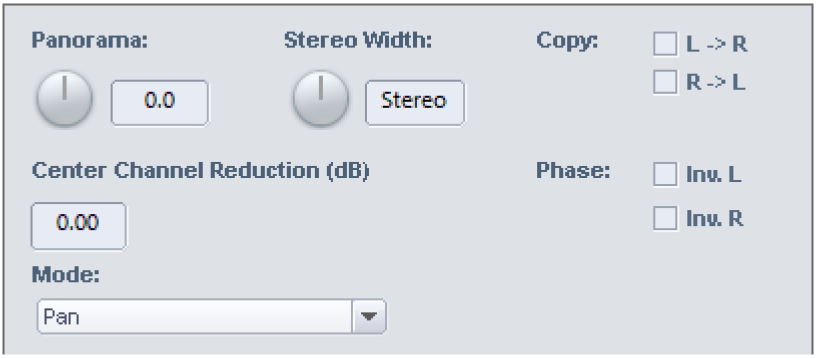
By right-clicking you can define the selected sidechain as a "direct out" (red fader display) or "post-fader out" (orange fader display), or open the stereo panorama dialog to set the panorama (view page 229) for the respective sidechain send.

A typical example for using a sidechain send is to control a compressor (view page 767) for the bass using the kick drum signal. This way, every time the kick drum sounds, the bass will be lowered.

You can get a good overview of all assigned sidechain sends in the matrix display of the Routing Manager (view page 196).

Stereo Panorama/Stereo Editor

Right-clicking on the panorama control in the arranger, mixer, track editor and object editor allows you to change the panning between the left and right channel for each track/object in the **Panning Dialog** dialog, as well as other settings that affect the stereo image and phase.



With mono tracks, the signal is converted from mono to stereo in the signal flow at the position of the panorama controller. Plug-ins or effects can still be added at this position. The routing positions can be changed in the Effects routing dialog (view page 677).

"Stereo Panorama" Dialog Options

Panorama: Here you can set level distribution between left and right.

Stereo width: Adjusts the stereo base width for each track. No change occurs in the middle position. Towards the left, the stereo signal is gradually replaced by the center signal (mono sum of L+R), thus reducing the stereo base. If the slider is set to the leftmost position, you will only hear the mono signal. To the right, the side signal (difference from L-R) is mixed in, thus increasing the stereo base.

> Detailed information about this is available in "Effects -> Stereo / Phase -> Multiband Stereo Enhancer (view page 836)" With the Multiband Stereo Enhancer you can edit the stereo bandwidth in three different bands frequency-selectively.

Copy: Here you can change the channels assignment. If you activate "**Copy L > R**", only the left channel will be audible on both sides; "**Copy R > L**" has the same effect for the right channel. Activate both options so that the left and right channels are swapped.

Center channel damping (panning law): In order to balance volume fluctuations while panning (particularly for automation) it is quite common to additionally dampen the track volume at the center position, while no additional damping occurs if the pan controller is moved to the left or right limit. The setting depends on the audio signal. Usual settings are:

- **0 dB**: This setting is recommended for stereo material. No damping occurs at the center position, i. e. if panned to the right, the right channel volume remains the same, if panned to the left the left channel remains unchanged. The audio material is not changed if set to the center position. For mono signals, this value may result in an increased volume if the signal is placed in the center.
- **-6 dB**: This setting can be used for mono tracks. If set to the middle, the level of the right and left channel is reduced by half.
- Common setting for automating stereo signals are **–3dB** or **–4.5 dB**.

Phase: Here you can invert the phase.

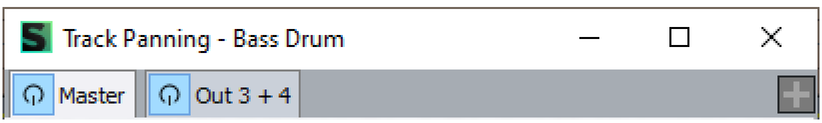
Mid-Side: With the option **MS->LR** you can convert a signal present as Mid/Side into a stereo signal. It is assumed that the mid (mono) signal source is on the left and the side signal is on the right.

Mode: Here you can determine whether the panorama should be calculated after - Pan - or before - Pan (StEnh post pan) - the stereo width. In addition you'll also find the options "2 Channel Panorama" and "2 Channel Volume":

- **2 Channel Panorama:** If you activate this mode, you can use both knobs to control the panorama of the left and right channels separately.
- **2 Channel Volume:** If you activate this mode, you can use both knobs to control the volume of the left and right channels separately.

The parameters are processed in the following order: Copy -> Stereo Width -> Phase Invert -> Panning/Panning Law

Additional options for track panning

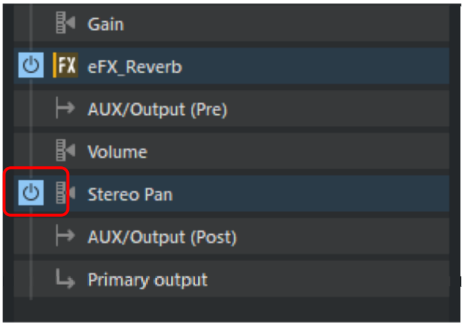


Use the + button to assign additional outputs (hardware or buses) to a track. The outputs can be switched on and off individually. For each of these outputs there is a separate panorama processing. This allows you, for example, to create a stereo and a surround mix in the same project. Click on one of the tabs to open the corresponding panorama editor. If the track is routed to a surround bus or master, the Surround Editor (view page 303) is displayed in the dialog accordingly. In the Stereo Editor there are some additional options:

All Tracks: Use this function to apply the panning law to all tracks/channels of the project. In the menu you can select whether the panning law should be applied to all tracks or to all tracks except buses. Also the option " Affects primary output only" (see below) can be transferred to all other tracks with this button.

Affects primary output only: If the track is routed to multiple outputs (e.g., stereo and surround master), and this option is active, the panorama is applied to the primary output only. This prevents panorama processing from being done twice if a track is routed to another output. In FX routing, panorama processing is moved to the very end of the effects chain, even behind the AUX/Output Post, the standard output for additional outputs.

This option can 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 "Middle/Side" and describes the process by which the channels are separated not into left and right, but into middle ("M") and side ("S").

MS recordings

With MS microphone setups files will be received after recording with separate middle and side parts, rather than a left and right channel. With the stereo panorama dialog's presets provided you can edit MS recordings (here we assume that these are available as a stereo file with M left and S right).

MS processing of mono original material

To extract the correct stereo picture from an MS file, duplicate the output material and save it as a new track. For the first track, select the "Left channel only" preset. This results in the M signal being played in mono only. On the second track, use the preset "Side signal (Stereo) (from M/S source)" to only play the side signal in stereo, i.e, left +side and right -side. Both tracks can be mixed together each at 0 dB

Tip: For this use case you can also use the **MS-> LR** option in the dialog.

MS processing of stereo original material

Another requirement is the independent editing of mids and side parts even if the output material is available in stereo format. To do this duplicate the output material and save it as a new track.

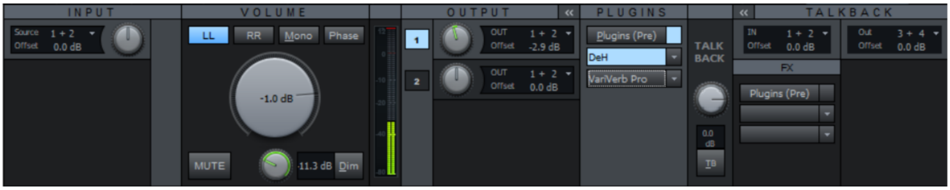
"**Mono (get mid signal from stereo source)**" extracts the mid part from the output material as the first track, while "**Side signal (stereo) (from stereo source)**" extracts the side part in stereo as the second track. Here you can subsequently mix both tracks each with 0dB.

Advanced Mid/Side processing

In Samplitude there are additional options for processing stereo signals in mono tracks.

- **Mono (get mid signal from stereo source):** In this case, only the mid component of the stereo signal is used in the mono track.
- **Left channel only:** In this case, use only the left component of the stereo signal used in the mono track.
- **Right channel only**: Use only the right component of the stereo signal used in the mono track.
- **Side signal (mono) from stereo source**: Use only the mono side component of the stereo signal used in the mono track. This way, you can apply all subsequent object and track effects to the side part as mono parts.
- **Side signal (stereo) from stereo source**: Use only the stereo side component of the stereo signal used in the mono track. This way, you can apply all subsequent object and track effects to the side part as mono parts.
- **Convert side signal (mono) to stereo** : Converts the mono side component to stereo.

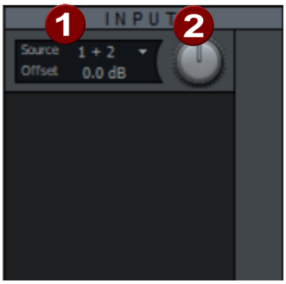
Monitoring section



The monitoring section provides a separate workflow for monitoring an input, bus or master. It is possible to switch between two different outputs, each with separate level settings and effect chains, for example to realize monitoring on different pairs of speakers (home system and studio monitors). There is also an independent talkback path for communication between the recording booth and studio.

> Note: Despite the similarity in its name to track monitoring (view page 72), this is another function independent of it. Track monitoring means listening to an input signal of a track, the monitoring section is used for general playback control of the program.

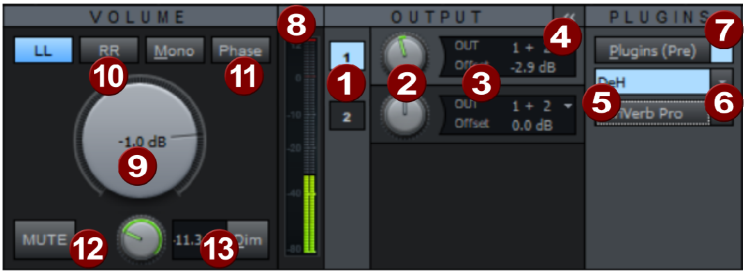
Input



1. **Select Source:** Default source is the Master, but you can also select any input pair or (stereo) bus as input source.
2. **Input Offset:** Here you can adjust the level of the input signal independent of the selected output.

> If the input is the Master or a bus whose output is routed to the same output device as the output of the monitoring section, the signal is not reproduced via both ways, but the monitoring section has priority.

Volume/Output/Plugins



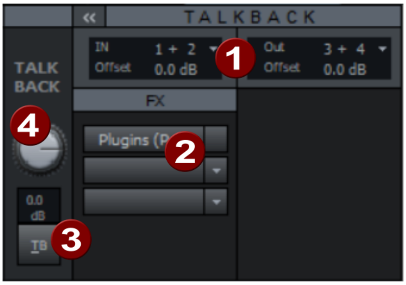
1. **Output selection:** A separate output device can be set for each output. All settings in the Volume and Plugins panels can be adjusted separately for both outputs.
2. **Output level offset:** The output level can be trimmed here per output.
3. **Output selection:** Select the output device here.

   The first output device corresponds to the device that can be set in the Monitoring section of the mixer (view page 222).
4. The plug-ins panel can be hidden with this button.
5. **Plug-in slots:** The plug-in slots work like those in the mixer, track or object editor. A left click on an empty slot opens the Plug-in Browser, a left click on a loaded plug-in deactivates the plug-in, a right click opens the plug-in window.

   You can even load more than two plug-ins, but only the first two are displayed in the slots. The other ones can be reached via the FX routing dialog.
6. Use the arrow to open the **plug-in menu**.
7. A left click on this button deactivates the entire plug-in chain, a right click opens the **FX routing dialog**.
8. The **Peak meter** shows the level behind the large volume control, but before the output offset.
9. **Volume control:** Here you set the monitoring volume.
10. **LL, RR, Mono:** Use these buttons to adjust the mono stereo signal distribution. With **LL** the left input signal is reproduced on both output channels, with **RR** the right signal. **Mono** sums right and left to a mono signal, which is then also reproduced on both output channels.
11. **Phase:** Inverts the phase of the right input signal to compensate for possible phase shifts.
12. **Mute**
13. **Dim:** A click on the button reduces the volume by the amount set on the knob.

Talkback

With the talkback function, you can set up an additional communication path between the recording booth and mixing console.



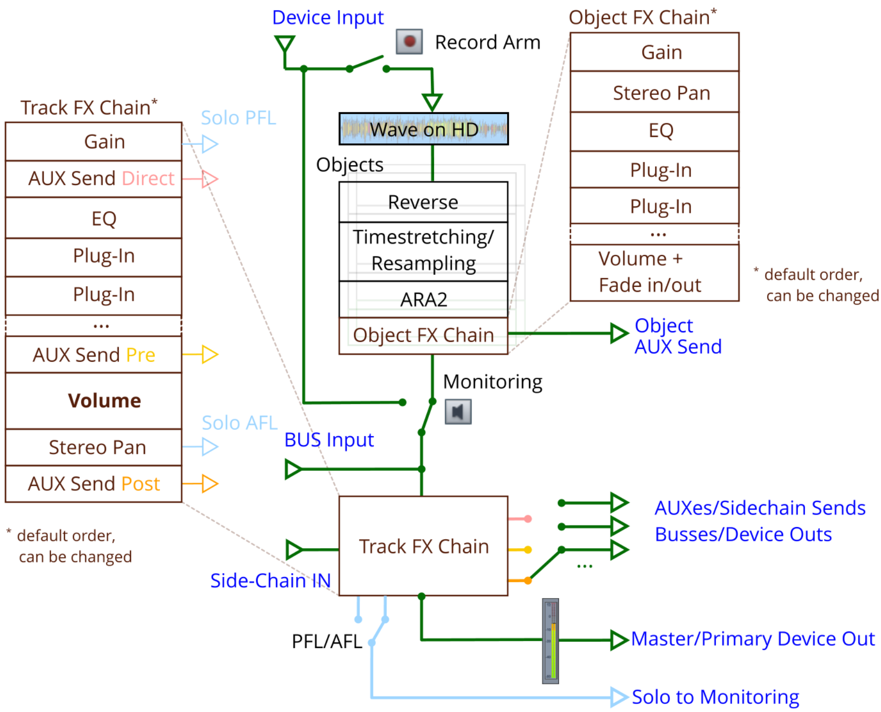
1. **Input/Output:** Here you define the input and output device for the talkback.
2. **Plug-ins:** The usual plug-in slots are also available for Talkback.
3. Activate talkback
4. Talkback Volume

Keyboard shortcuts for the Monitoring section

To be able to execute the most important monitoring functions quickly without mouse clicks, there are some keyboard shortcuts especially for the monitoring section. These only apply when the monitoring window is open and focused. To use them, you must therefore have clicked into the window at least once after opening it.

		H	Invert phase
1, 2	Select Output 1 or 2	**B**	Open Plug-in browser (Monitor)
Alt+M, 0	Mute	**Shift + B**	Open Plug-in browser (Talkback)
D	Dim	**P**	Plug-ins on/off (Monitor)
L	LL (Input L to Output LR)	**Shift+P**	Plug-ins on/off (Talkback)
R	RR (Input R to Output LR)	**T**	Talkback (momentary)
Alt+N	Mono (Input L+R to Output LR)	**Shift + T**	Talkback on/off

Mixer Signal Flow



Remarks:

- A track can have any number of AUX send, sidechain send or output routings to busses or hardware outputs, for each routing one of the three positions in the track effect chain can be specified.
- The Record Arm and Monitoring switches are dependent on each other depending on monitor switching behavior (view page 79) and playback state.
- The bus input of a track is only active when the channel operates as an AUX/submix bus and always receives the sum of all aux sends or routings of the other tracks, regardless of the playback state.

Effects – Organization and Workflow

Internal real time effects

During effects editing, effects are calculated during playback or export in real time. The effects settings can be changed at any time. Real-time effects can be applied to different positions in a project:

- **Object effects** (access via the object editor or the "Effects" menu)
- **Track effects** (accessible via "Track" > "More" > "Track effects", the track editor, track header plug-in selection field or the insert section of the mixer)
- **Master Effects** (can be accessed via the Master section of the Mixer)

The signal passes through the real-time effects in the following order:

1. Object effects
2. Track effects
3. Master effects

Internal offline effects

The object effects can also be added as offline effects. To do this, activate "Process effects offline" in the "Effects" menu.

In this editing process, effects are added on a one-time basis to the original wave file or a new wave file. In contrast to real-time effects, no additional CPU resources are required during playback of the effect.

You can switch between non-destructive and destructive offline audio editing in the effects dialogs ("Create copy" option):

- **Non-destructive audio editing:** The effect for the selected area will be written into a temporary file. When saving the WAV file, the original and the changed areas will be recombined again.
- **Destructive audio editing:** The effect for the selected area will be written directly into the original file. The "Undo" function is available for reversing operations.
- **Application on a VIP object:** The effect will be calculated for the selected area of the object. Depending on the settings for the application of destructive effects, calculation takes place directly in the original material, at the end of the original material or in a new file.

To see precise settings for effects editing, go to "Destructive effects editing" in the system settings ("Y" key).

Effects plug-ins

Along with internal effects, additional effects can also be applied as plug-ins. Samplitude supports plug-ins in the following formats:

- **MAGIX plug-ins**: A selection of internally integrated VST effects that come with the program. Includes the Analogue Modelling Suite (only for Samplitude Pro X6 Suite and Sequoia), the Vintage Effects Suite, AM-Munition, essentialFX (eFX), Vandal, and VariVerb II.
- **VST effects-**: This includes all external VST effects

These effects can be accessed in the track editor, object editor, track header or via the "Insert" section of the mixer, the routing dialog and the master section.

Effects - Signal flow

Offline effects are applied before all realtime effects because they are immediately calculated into the audio material. Real-time effects on the other hand, do not change the original audio material. They are calculated during playback in "real-time".

The signal passes through the realtime effects in the following order:

1. Object effects
2. Track effects
3. Master effects

Effect slots

Effects are loaded to their location via the effect slots.

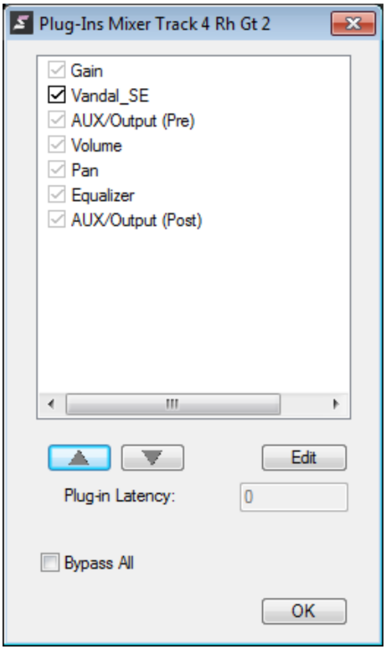
Effect slots in the object editor, track editor, mixer and track header

Clicking on the empty insert slot opens the plug-in browser (view page 240), where you can load a plug-in to the slot. Click on an occupied slot to activate/deactivate the plug-in. Right-clicking on the slot opens the interface of the plug-in. The arrow next to the respective insert slot opens a menu with various functions. You can ope the plug-in browser again to switch out the plug-in or remove it ("No effect"). The "Plug-ins" button activates or deactivates all effects in the channel. A visual indicator (*) is displayed for plug-ins that were previously active and are active the next time "Plug-ins" button is pressed.

Effect Routing Dialog

In the Effect Routing/Plug-ins dialog you can make all the important settings for realtime effects and plug-ins. This dialog is available at object level ("FX" in the object editor), channel/track level ("FX" button in the track editor), and at mixer level ("FX" button in the stereo channel strip).

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.



1. When you move the mouse over an element, controls for that element become visible:

   Deletes the current entry from the effect chain.

   Displays the user interface of the effect. For AUX outputs the AUX send routing dialog (view page 226) is displayed, for Pan the corresponding Panorama dialog (view page 229).

   The arrow keys move the element in the chain.

2. Use the arrow keys to move selected effects in the chain.

3 The entries **AUX (Direct/Pre/Post)** indicate the points in the effects chain where the signal for the AUX send busses is tapped. As with physical mixers, these are preset directly before and after the volume, but they can be moved just as freely as the effects.

You also define the positions for additional track output routings (More information can be found in the section "Assign track to multiple outputs" on page 212).

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.

4 **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. These settings can be accessed via the menu on the FX button at various points in the program.

7 **Plug-In Latency/Force Latency:** Samplitude works with latency compensation for all plug-ins. Plug-ins report their latency to Samplitude and this value is transformed into a time shift in the audio material. Here you can read the latency reported by the plugin.

8 Click **OK** to close the FX routing dialog.

Plug-in browser

This function has been revised in Samplitude. For the latest information, see the PDF document **Samplitude Pro X7 New Features** in the program folder.

The plug-in browser or FX browser allows all instruments and effects to be loaded to the slot at object, track and master levels. It offers centralized access to all available plug-ins and instruments.

1. **Full text search:** After opening the FX browser, the input field is selected for the full text search. Simply start typing and enter the name or part of the name of the plug-in you're looking for. The search function filters the plug-in list below and selects the first search result. You can use the enter key to load the selected plug-in to the track slot.

2. **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.

3 **Plug-in list:** All plug-ins in the list and optionally their presets are listed in a tree structure according to setting under **4.** Using the arrow keys you can move through the list to select a plug-in. Ctrl + Arrow left collapses the entire tree and Ctrl + Arrow right opens up the entire tree.

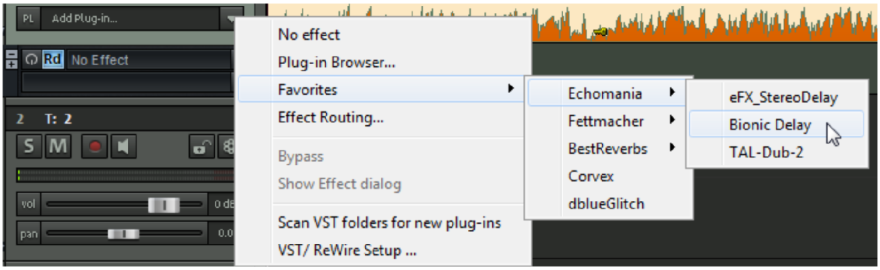
4 **Instruments/Effects:** The display can switch between instruments and effects. Instruments are only used at track level and can therefore only appear there.

5 **Path/Manufacturer/Category:** The tree structure can be sorted according to storage location, plug-in manufacturer or plug-in category.

6 **Last used:** This view lists the previously used plug-ins.

7 **Favorites:** You can mark your favorite plug-ins and presets with a star. If the "Favorites" button is activated above, only your favorites will display in the list.

Double-click on "New folder" to create folders and double-click on the name of the folder to rename it. Drag & drop to sort your favorites into folders – you can also move a folder within another folder. Favorites are accessible via the plug-in slot menu and the folders are displayed there as sub-menus.



8 Here you can see whether a plug-in is VST2 or VST3, and whether it's a 32-bit or 64-bit version.

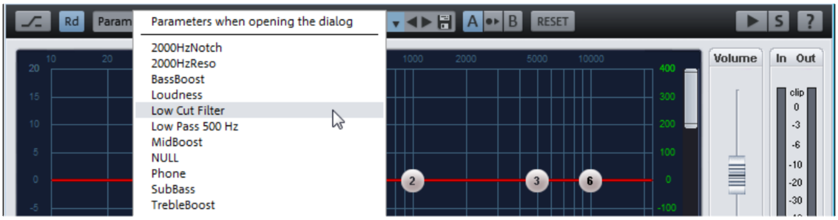
9 **Add:** Loads the effect or instrument into the effect chain, opens the interface and closes the plugin browser. The opening of the effect dialog and the closing of the plug-in browser after adding the plug-in can be deactivated with the respective option.

10 **Options:** Opens a menu with settings. You can scan for new plug-ins or presets and open the VST settings (view page 615).

11 **Effect routing:** This displays the Effect routing dialog (view page 239) of the track, master or object. In this you can change the effect order, display or delete effects.

Saving Effect Parameters (Preset Mechanism)

Most of the effects include preset selection boxes. Presets appear in the selection box when they are moved to or saved in the folder "Program Files" > "MAGIX" > "Samplitude" >fx-preset".



If a preset is not saved in the "fx-preset" folder, you can open it using the "Load" function. However, when opened this way the presets will not appear in the selection box. In this case the path to effect preset has to be set manually.

Effect Dialog Bar

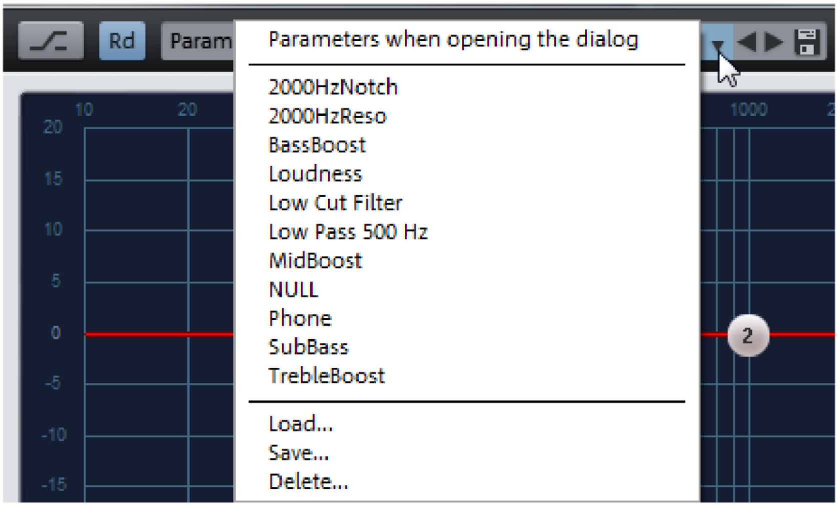
In several effect dialogs you have direct access to the automation button which also provides access to the automation menu. This is made possible by a newly created header in the respective dialogs.

In the presets input field you can load, save or delete settings.

By selecting "Parameters when dialog is opened", you will undo all changes made in the dialog since it was opened.

By closing the dialog, you will apply all current settings.

Clicking on the downward pointing arrow, you will see all presets at a glance. Below the preset list you can open dialogs for loading additional presets and for deleting selected presets. You can navigate through presets using the two arrows pointing in horizontal directions. Current settings are saved to disk.



Here is an overview of the other control elements:

Bypass: The effect is removed from the signal route. This enables the unedited signal to be easily compared to the result of the current effect settings.

Mode: By clicking on this button you can toggle between the selected automation write/read mode. By right-clicking you can get to the Automation context menu.

A/B comparison:This control element is only displayed when it is supported by an effect. You can copy each selected setting using the arrow symbol to a different location. This way you can experiment with successful settings without losing them.

Reset: This control element is only displayed when it is supported by the respective effect. All parameters are reset to default settings.

Play button.

Solo button. This control element is only displayed when it is supported by the respective effect. The Solo mode will be activated for the corresponding effect.

Help: You can get extra information in the help section.

Track effect settings

By right clicking on the track/channel number in the Track Editor, Header or Mixer a Context menu will open. In the "Track effects" category you can copy, reset, save or load current/discarded track effect settings. There are also a wide range of presets available, which offer you possible effect combinations and instruments for various applications.

The track effect settings are .TRK files. Save your personal track effect settings in the program directory in "FX presets" -> "Track FX". You can also create subfolders here. We've already included a selection of useful presets, e. g. for "Mid-side processing". The track effect settings of a VST instrument can also be saved (including parameters and all subsequent track effects), and transferred to other tracks.

Effect Routing

The sequence of all realtime effects can be adjusted individually for each track. This enables individual sequences to be set for the effects.

> More information about this can be found in the chapter „Mixer“ > „Effect Routing/Plug-ins Dialog (view page 239)“.

External hardware effect integration

> This function has been revised in Samplitude. For the latest information, see the PDF document **Samplitude Pro X7 New Features** in the program folder.

(for "Mixer FX Monitoring/Hybrid Engine" only)

To integrate external effects devices into your setup, you will require a multi-channel audio interface. Reserve a channel pair for every external effect device you want to integrate.

The integration of external effects or synthesizers can be done in the dialog "File > Project Properties > External Effects". Here you can set up the inputs and outputs for 32 external devices, create new "Effect send" and "Effect return" tracks, as well as specify effects latencies so that they can be taken into account during latency compensation.

The order of inputs and outputs for the program is saved globally. The setup of external effects integration is project-specific.

Note: Return tracks have to be recorded first in order to be available for track bouncing and CD burning. For this reason, external effects are integrated through the program's own tracks and are not available in the tracks themselves as plug-in inserts.

You can open the "External Effects" dialog in the project options or by using the keyboard shortcut "I".

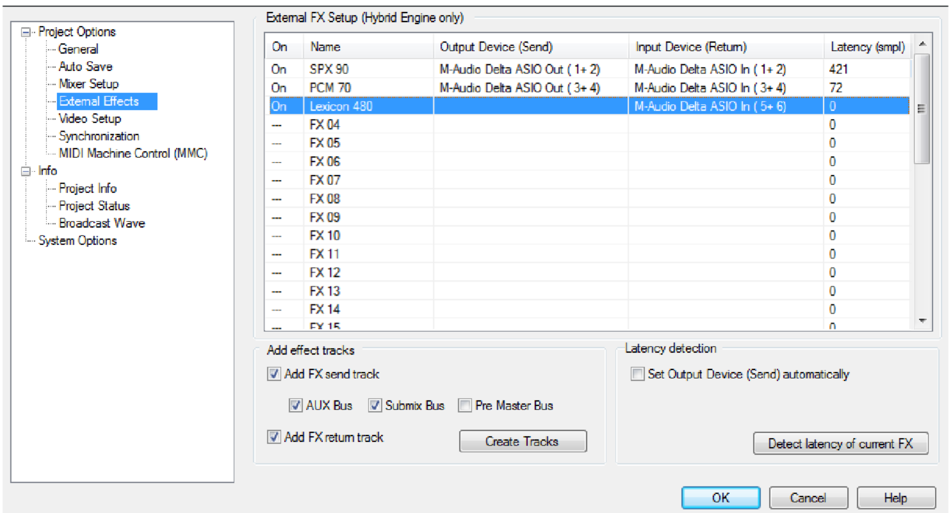
Routing Effects

Step 1: First replace the descriptions FX01, FX02 etc. in the name column with the name of the effect that will be inserted.

Step 2: Clicking on the next column reveals a selection of available output devices that can be used as effect sends for the effect device. Select an output.

Step 3: In the „Input Device“ column select a free input as the effect return channel.

Step 4: Now you can activate the external effect device by clicking in the first column and connect it to the set physical audio inputs and outputs.



In all areas of the program where input and output channels are set up, you can also integrate external effect devices, e.g. in the mixer, in the track editor or by right-clicking on "Mute" or "Record". The names of the assigned external effects are displayed behind the device name.

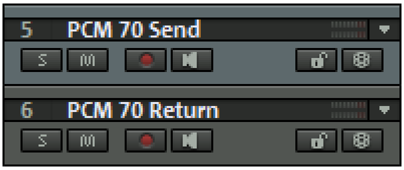
Creating effects send and effects return tracks

The lower part of the "External effects" dialog lets you create effect tracks for your external effect device.

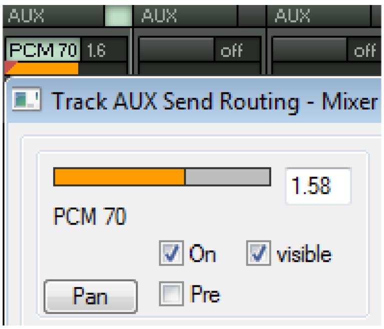
Effect as AUX

Step 1: Under "Add effect tracks", select the options "Add FX Send track" and "AUX bus", then tick the box beside "Add FX return track".

Step 2: Click the "Create tracks" button. You'll now see two newly created tracks or channels in the arranger and in the mixer that control the send and return signals of the external effect device.

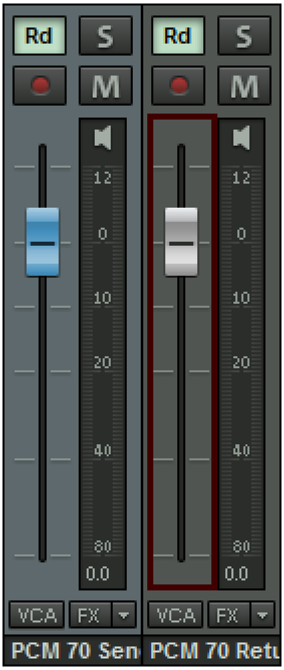


Step 3: You can now set up the send level of every integrated effect device in the AUX slot of every mixer channel



The volume fader of the send channel works as master send level.

Step 4: Position the effect signal in the mix using the fader of the effects return channel.



Effect as Insert

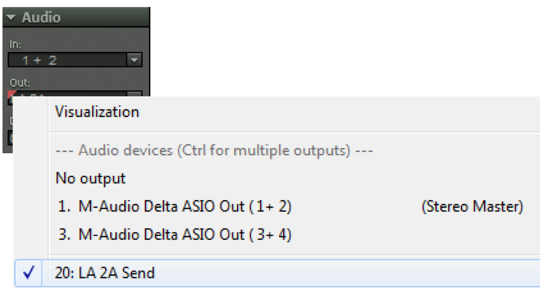
To add an insert effect:

Step1: Select "Add effects track", then the options "Add FX send track" and "Submix bus", then set a check next to "Add FX return track".

Step2: Click "Add tracks".

Two newly created tracks will now appear in the arranger and in the mixer to control the send and/or return signal of the external insert effect device.

Step 3: For the track that will receive the insert effect, select the Send (Submix) track to the external effect device as the output device.



Step 4: With the fader of the effect return channel you can set the return level of the effect signal.

Note: Make sure input monitoring is switched on for the effect return track. This can be done by clicking on the track monitoring button (loudspeaker symbol). The loudspeaker symbol is highlighted and you will hear the effect return signal.

The routing for each channel can be set up in the track settings dialog:

Step 1: Right-click on the track name.

For the effect send channel, set the output for your sound card that is connected to the external effect input in the "Playback" field.

Step 2: For the effect return channel, set the output for your sound card that is connected to the external effect output in the "Record" field. For the channel that is intended to work with the external device, set the effect send track in the "Playback" field [e. g. Submix: LA 2A Send (track 16)].

Use the double arrow buttons beside the track name field in the track options dialog to switch the track options to those of the neighboring track.

It is also possible to use an effect as an AUX as well as a submix bus:

To do this, activate both options before creating new tracks. FX sends can be changed later by accessing the track properties (right-click on the track header for the FX send bus) and setting checkmarks next to the corresponding settings "AUX bus" or "Submix bus".

Effect as pre-master bus

If you select "Pre-master bus" when creating tracks, the new tracks will be added at the end of the project, and all project tracks will be routed to the input of the send track. The return track is automatically routed to the master. This enables the external effect to be used as a master effect.

Setting latencies for external effects

Almost every external device features latency that needs to be compensated to function synchronously with the tracks in the arrangement.

Use the "**Set effect latency**" button to set the latency of an effect. A ping signal is transmitted to the "Send", and the time it takes to reach the "Return" channel is determined by this signal. The resulting value features a doubled ASIO buffer size and the latency of the external effect device.

The option "**Automatically identify output (send) device**" emits ping signals to all available devices. The device that matches the effect is detected automatically. The input device for the effect must be set prior to this.

Integrating external synthesizers

Select the MIDI port as the output device (send) to control the synthesizer here. Specify the input of your sound card where the synthesizer is located as the input device (return). Press the "**Set effect latency**" button to set the latency of the external synthesizer to ASIO buffer size.

Effects and Plug-ins in an Overview

Realtime Effects at Track, Object, and Master Level

Amplitude: Normalize, Normalize (Quick Access), Loudness Normalization, Fade In/Out, Set Zero

Dynamics: Advanced Dynamics, Multiband Dynamics, sMax11, efx_Compressor, eFX_Gate, AM-munition (Samplitude Pro X6 Suite), AM-track (Samplitude Pro X6 Suite), AM-phibia (Samplitude Pro X6 Suite), AM-pulse (Samplitude Pro X6 Suite)

Frequency/Filter: Parametric EQ, EQ 116, FFT Filter/Spectral analysis, Brilliance Enhancer (optional), Filtox, eFX_DeEsser.

Delay/Reverb: Delay, Room Simulator, efx_Reverb, efx_StereoDelay, Ecox, Variverb Pro.

Time/Pitch: Resampling/Time stretching, Elastic Audio.

Distortion: Distortion, Vandal, Bit Machine, eFX_TubeStage

Restoration: DeClicker/DeCrackler SE, DeClipper SE, DeHisser SE, DeNoiser SE, Get Noise Sample, eFX_DeEsser, Remove DC Offset (offline)

Stereo/Phase: Switch Channels, Multiband Stereo Enhancer, Invert Phase (both channels, left channel, right channel), eFX_TremoloPan

Modulation/Special: Vocoder, eFX_ChorusFlanger, eFX_Phaser, eFX_VocalStrip, Corvex, Reverse

Sample Manipulation: Adjust Sample Rate (offline), Reverse, Form Loop (offline)

Plug-ins:

essential FX: eFX_ChorusFlanger, eFX_Phaser, eFX_Reverb, eFX_StereoDelay, eFX_Compressor, eFX_Gate, eFX_Limiter, eFX_DeEsser, eFX_VocalStrip, eFX_TubeStage, eFX_TremoloPan.

MAGIX Plug-ins: AM-Munition (Samplitude Pro X6 Suite), AM-Track (Samplitude Pro X6 Suite), AM-Phibia (Samplitude Pro X6 Suite), AM-Pulse (Samplitude Pro X6 Suite), Corvex, Ecox, Filtox, VariVerb Pro, Vandal

Process Only Left Stereo Channel

Process Only Right Stereo Channel

Apply effects offline: Tick the checkbox here to edit effects offline

At object level you can swap object channels, invert phases and open the Object EQ, Object Dynamics, Pitch shifting/Time stretching, Object backwards, Elastic Audio, and MAGIX plug-ins.

Cleaning/Restoration Suite (optional)

With the optional "Cleaning/Restoration Suite" the following cleaning effects are available as real-time effects and as offline effects:

- **DeClicker/DeCrackler:** The DeClicker removes crackling and clicking noises, which are typical on scratched records.
- **DeClipper:** The De-clipper removes overmodulation
- **DeHisser:** The DeHisser eliminates regular, "white" noise, typically produced by recordings, microphones, pre-amplifiers or transformers.
- **DeNoiser with Noise Print Wizard:** The DeNoiser removes annoying background noise from audio material.
- **Brilliance Enhancer:** The included Brilliance Enhancer compensates for the loss of higher frequencies, for example due to MP3 compression or in the case of old tape recordings.
- **Spectral Cleaning:** This removes distortions (like coughing, whistling, or singular claps) from a recording without influencing the wanted signal. The graphical interpretation of the music can be seen in the Spectral Cleaning Editor via a spectrogram.

Once you've bought the Cleaning/Restoration Suite effects DeClicker/DeCrackler, DeClipper, and DeNoiser and Spectral Cleaning you can find them in the **Effects menu** under "**Restoration"**.

Once you've bought the Brilliance Enhancer you can find it in the **Effects menu** under "Frequency/Filter."

VST effects

VST-compatible plug-ins can also be used for processing effects in Samplitude. This allows almost any effects algorithm and VST instrument from third-party vendors to be used in addition to the effects included in Samplitude.

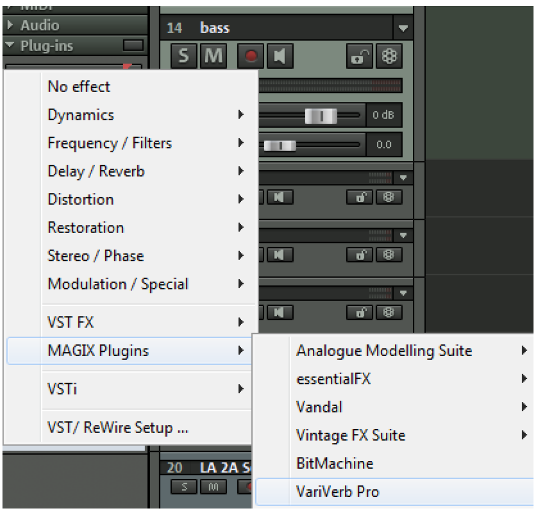
VST Instruments

In the track editor the MAGIX synths DN-e1, Revolta 2 and Vita as well as the Independence Sampler, Vita Solo Instruments, a list of all integrated VST instruments, the object synths BeatBox2, Loop Designer and Robota plus all devices connected through ReWire are all accessible in the first plug-in slot of each track. You can include your VST plug-in folder via "File > Program Preferences > Effects > VST/ReWire" in Samplitude.

Detailed information about plug-ins can be found in "Software/VST plug-ins/ ReWire (view page 411)".

MAGIX Plug-ins

MAGIX plug-ins are **effects** that are accessible via the track editor plug-ins, object editor plug-ins, track header plug-ins selection menu, track header plug-in button, the insert section in the mixer, or via the "Effects" menu. These are further divided into the **Analogue Modelling Suite (AM-Phibia, AM-Pulse, AM-Track) (optional)**, **Vintage Effects Suite (CORVEX, ECOX, FILTOX), essentialFX (eFX_ChorusFlanger, eFX_Phaser, eFX_Reverb, eFX_StereoDelay, eFX_Compressor, eFX_Gate, eFX_Limiter, eFX_DeEsser, eFX_VocalStrip, eFX_TubeStage, eFX_TremoloPan), VariVerb Pro, AM-Munition (optional)** and **Vandal**.

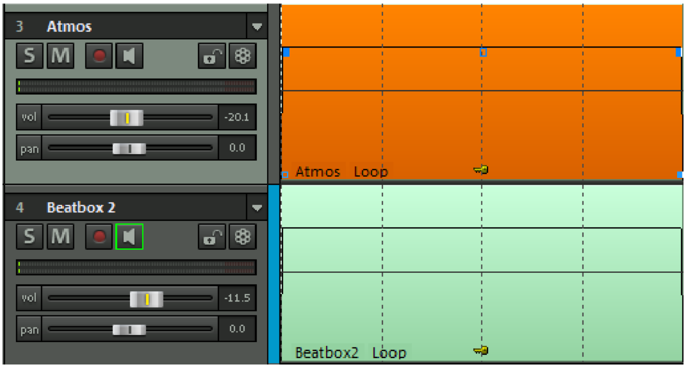


Object Synths

The Object Synths feature a particular form of synthesizers which form individual synth objects in Samplitude. You can move synth objects between tracks any way you want without affecting the synthesizer settings in the process. In this way, they differ from other software synthesizers, which are track-dependent and operated via MIDI like MIDI synthesizers.

The Object Synths are available as installation options. After installation they can be found in a special "Synth" folder within the program folder.

Synth Objects can be accessed through "Object -> New Synth Object". This creates a 4 bar loop object at the current playback marker position. The instrument interface for the synth object that is created can be accessed by double-clicking on the object.



The keyboard shortcut "Ctrl + Spacebar" plays back the selected synth object in "Solo" mode.

The following object synths are integrated into Samplitude:

- **BeatBox 2**, a virtual drum computer for drum kits with high quality effects
- **Loop Designer** for developing your own drum loops and bass lines
- **Robota** a complex, four-part drum computer that aids in the generation of sounds as well as working with samples and oscillators (analog sound synthesis).

BeatBox 2

BeatBox 2 is a 16-voice pattern-based drum computer featuring hybrid sound synthesis and a step sequencer. The interface includes a multi-effects section (one effect per drum instrument), auto copy, and a convenient velocity editing function.

A drum sound is created in BeatBox 2 using a sample, which is combined with a synthetic sound that can be created using up to three different synthesis models (hybrid sound synthesis). BeatBox 2 also enables detailed editing and automation of all sound parameters.

BeatBox 2's interface uses two states. While closed, you can listen to the included sounds and patterns or those you've made in BeatBox 2 without using too much window space.

Only the most important control elements are displayed when the program window is minimized:

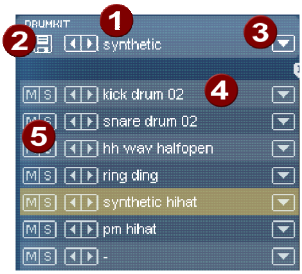
1. **Volume controller**: Controls the volume.
2. **Peak meter and preset name**: The peak meter enables the output from BeatBox 2 to be checked visually. Clicking the triangle beside the preset name opens the preset list.
3. **Previous/next preset**
4. **Save preset**: The preset includes the drum kit in use, the pattern, and any possible automations (see page 249).
5. **Play/stop**: The playback controller in BeatBox 2 starts the BeatBox solo, i.e. without playing the arrangement.
6. **Edit button**: The edit button opens BeatBox 2 for you to program your own beats and sounds.

Maximized BeatBox 2 plus window:

1. **Drum kit**: This section is where drum kits (collections of different drum instruments) and the individual drum instruments are loaded.
2. **Selected drum instrument**: The settings in the synthesis section (5) and velocity/automation (4) always affect the selected drum instrument.
3. **Pattern editor**: Programs the beat sequence. In the top part, different patterns (sequences) may be loaded and saved as well as different settings for the view and function of the pattern editor. The matrix is where the beat is programmed: one line corresponds to a drum instrument, and a column matches a specific time position within 1-4 beats. If a cell is clicked, then the respective drum instrument will be triggered at this position.
4. **Velocity/control**: This section has two modes: velocity and automation. The "Velocity" setting displays the velocity levels for the selected instrument's beats as a column. The "Automation" setting allows automation to be set for sound parameters in the synthesis section (5).
5. **Synthesis**: The selected sound parameters and effects settings for the selected drum instrument may be edited here.

The following section explains the sections of BeatBox 2 individually:

BeatBox – drum kit



This section loads drum kits (collections of different drum instruments) and the individual drum instruments. You can also try out an already programmed pattern with different kits or exchange individual drum instruments.

1. **Select drum kit**: Use the <> buttons to cycle through the different drum kits. A drum kit is a collection of percussion instruments with matching sounds, e.g. rock kit or electronic drums à la TR 808. By changing the drum kit, you can add an entirely different sound to the rhythm you have already created.
2. **Save drum kit**: Use this button to save the current collection of drum instruments as a kit.
3. **Drum kit list**: Click the arrow to the right of the name to open a complete list of available drum kits.
4. **Select drum kit**: The arrow buttons function the same way as with the drum kit. The sequence of drum instruments in the kit may be resorted via drag & drop.

5. **Mute/solo**: The "Solo" button switches a drum instrument solo, i.e. all other instruments which are not "solo" will be muted. The "mute" button mutes a drum instrument.

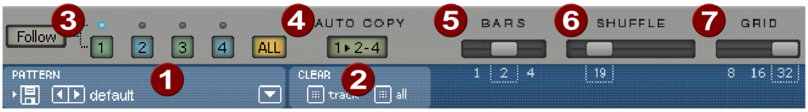
New drum or effect sounds may by added to the current drum kit via drag & drop from Windows Explorer. Drag a wave file to a drum instrument to create a new drum sound based on this sample. BeatBox 2 copies the sample into the sample folder to make sure that the instrument or kit created can be used again later. Drag a complete folder featuring wave files to the drum kit to create a kit based on those samples.

BeatBox – context menu

Right clicking a drum instrument always a context menu:

- **Copy/paste**: Copy an instrument from a track and paste it to another one.
- **Empty instrument**: An empty instrument is added. No sound is played, it has no name, and is used to clean up an unused track.
- **Default instrument**: Adds the standard instrument. This features the standard parameter for all synthesis shapes and serves as the starting point for your own sounds.
- **Reset automation**: Several of BeatBox 2's own presets contain automations. These are dynamic sound parameters like filters or pitch changes. This command allows these to be completely removed from the selected instruments.

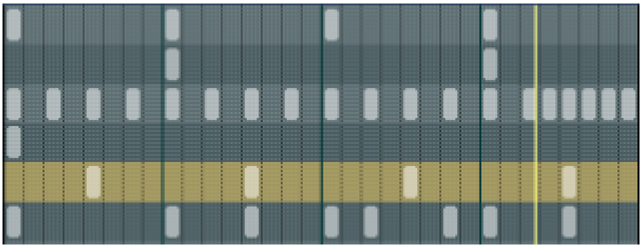
BeatBox – pattern editor buttons



1. **Pattern**: Use the <> buttons to switch through the different patterns. The arrow to the right opens a list of all available patterns, and the save button stores the current pattern.
2. **Clear track/all**: All events for the selected instrument (track) or all events for the pattern (all) are removed by clicking this button.
3. **Bar selection**: The bar you wish to edit may be selected via the corresponding number button. Use the "Follow" button to select "Follow" mode, i.e. the step display follows the steps of the currently played beat. "All" displays all bars of the pattern.
4. **"1>2-4" auto copy**: If more than one bar is set as the pattern length, "Auto copy" mode ensures that the drum notes set in the first bar are automatically placed into the next bars. This also makes it easy to create a continuous beat, even with a loop length of four bars. Notes set in the bars further back are not affected by the "Auto draw" function, e.g. making faint variations detectable only in the fourth bar.

5. **Bars**: The maximum length of a drum pattern is four beats. This length may be selected via the small scroll bar.
6. **Shuffle**: This controller changes the timing of BeatBox 2; if the controller is turned to the right, 1/8th notes in the rhythm will be played more and more as triplets. If that sounds too abstract, simply try it out, ideally with a pure 1/16 hi-hat figure; you'll soon see what the shuffle fader is capable of.
7. **Grid**: Set the time resolution for BeatBox here. Choose from 1/8th notes (only for very simple rhythms), 1/16 (default), and 1/32 (for more refined constructions).

BeatBox - Pattern editor Matrix



This is the heart of BeatBox. With a mouse click on any position in the drum matrix you can create and delete any drum note. Clicking and dragging draws in a series of notes. Together with the velocity editing options (see Velocity), you can easily create drum rolls.

If "Shift" is held when you click on notes in the range, a rectangle can be drawn out which selects the notes contained within this rectangle (lasso selection). Selected notes can be copied by dragging them to a new position. If "Ctrl" is also held down, then existing notes will remain at the target position. Delete all selected notes by right clicking.

Two special commands are available for selection:

Shift + double click: Select everything in the beat that has been clicked.

Ctrl + shift + double click: Select everything

A simple mouse click cancels the selection. The selection is automatically canceled after copying. If you want to keep your selection, hold down the Shift key while copying.

Many functions in BeatBox 2 can be controlled with the keyboard. For example, a beat can be triggered with the "Enter" key live in a running pattern. Here's a complete list of the keyboard commands:

General

E	Open/Close editor

Pattern editor options

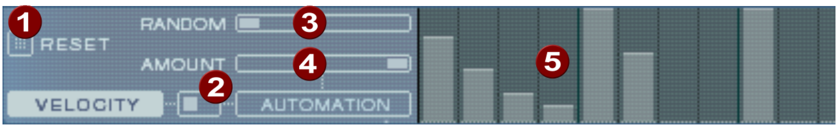
A	"1>2-4" Auto copy
F	Follow
1-4	Display bar 1... 4
0	Show all bars
+/-	Grid finer/rougher

Selected drum instrument

Arrow up/ down	previous/next
P	Preview
Enter key	Live input
M	Mute on/off
S	Solo on/off

BeatBox – velocity

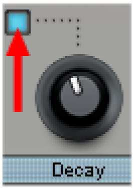
Use "Velocity" mode to edit the velocity/automation of the individual drum notes for the selected drum instrument.



1. **Reset**: Sets all velocity levels to 100%
2. **Mode switcher**: Switches the range between velocity control and automation (see page 249).
3. **Random**: The random parameter adds random variations to the set velocities. This helps make your beats sound more natural.
4. **Amount**: No function in velocity mode.
5. **Velocity level**: Enables every set note for the selected instrument to be edited via the height of the column. Multiple columns may be edited at once; see "Editing velocity values and automations".

BeatBox – automation

Every parameter for a drum sound, including effects, may be automated, i.e. so that it changes during a pattern. For example, make your snare drums more lively by adding higher voices to loud hits or by setting accents on individual hits with reverb effects.



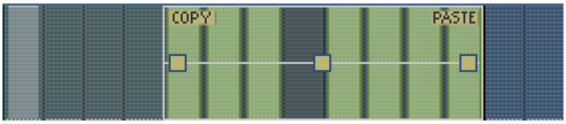
The synthesizer section below enables the selection of a parameter for automation via the small blue LEDs above a parameter controller. More about the parameter controllers is available in the "Synthesizers" section.

1. **Reset**: Sets all automation values for the selected parameter to "0".
2. **Mode switcher**: Switches the section between velocity control and automation. This happens automatically when selecting parameters.
3. **Random**: The random parameter adds random variations to the set automation. This helps make the beats sound more natural because each programmed hit will sound slightly different. The level of the random parameter is also influenced by the amount controller (see below), i.e. if the amount is at "0", then the randomness will have no effect.
4. **Amount**: The amount parameter regulates the complete influence of the automation values and of the random factor on the selected parameter. An amount of "0" does not have any effect at all on the automation; maximum level has the greatest effect. The effect of the amount controller on the automation values is made visible by a slightly brighter line in the value columns.
5. **Automation values**: The selected parameters may be drawn in as a column with the mouse. Automation values may also be drawn between the set notes; the sound of the drum instrument changes during playback. The automation values are added to the original parameter value.

Editing velocities and automation values

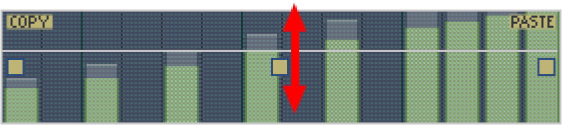
Hold down "Shift" and select any number of bars for velocity or automation with the mouse. Two special commands are available for selection:

Shift + double click	Select everything clicked on in the bar
Shift + Ctrl + double click	Select all

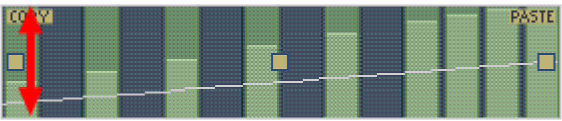


The "Copy" button copies the selection to the clipboard. If you select this or that track in another editor, you can paste the notes there now from the clipboard. If the target selection is larger than the contents of the clipboard, then it will be inserted again. This lets you quickly add a short section throughout the complete length of the pattern.

The three handles allow the velocity or automation values to be edited together.

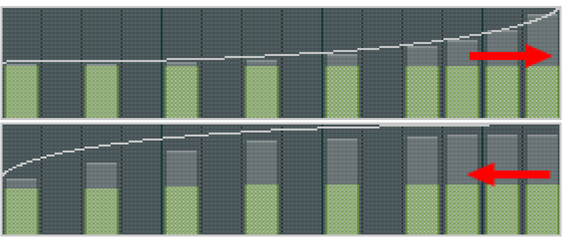


The middle handle increases or lowers the values together.



If the handles are moved horizontally, you can change the curve shape of the transition.

A single click in the automation section cancels the selection again.



If you move the handles horizontally you can change the curve shape of the transition. A single click in the automation section cancels the selection again.

Note: A selection of velocity values matches the selection of corresponding notes in the matrix editor track.

BeatBox – synthesizers

Use the lower section of BeatBox to set the sound for the selected drum instrument.

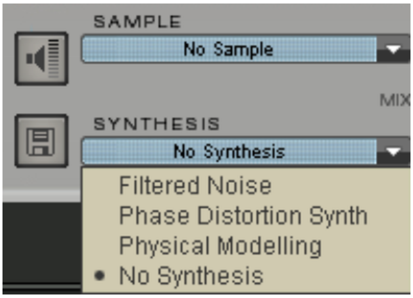
Synthesis in BeatBox 2 consists of a combination of a simple drum sampler and a synthesizer. There are three different synthesis models possible for the synthesizer: "Phase distortion synth" (FM synthesis), "Filtered noise", and "Physical modeling". The mixed signal of both components is then processed by a multi-mode filter. An envelope curve ("envelope generator") time-dependently controls modulations of in all components.

1. **Preview drum instrument**
2. **Save drum instrument**
3. **Select sample**: Clicking the arrow selects samples from the categories like kick, snare, etc.
4. **Select synthesis mode**: Switches between the three synthesis models.
5. **Mix**: Mix ratio between drum sampler and synthesizer.
6. **Parameter controller**: All six sound parameters for a drum sound may be set directly and automated via the parameter controllers. These parameters depend on the currently loaded drum sound. By clicking the name of a sound parameter, a menu opens for you to add parameter controllers to each drum sound's synthesis parameters.
7. **Automation**: This selects the controller's parameter for the automation.

BeatBox – synthesis models

The sound synthesis in BeatBox 2 consists of a simple sampler and a synthesizer which includes three different synthesis modules.

Sampler: The sampler plays short sound recordings ("samples") in different pitches. The sampler is intended for creating all kinds of drum sounds; the sounds are static and unnatural if the pitch is not altered. This is why the sampler is combined with on of the three synthesis models.



Filtered noise: White noise is filtered by two band-pass filters with separately adjustable frequencies and resonance. This algorithm is suitable for creating synthetic snares and hi-hat sounds.

Phase Distortion Synth: Two oscillators with regulated phase distortion and extremely variable frequency modulate each other (FM/cross-modulation/ring-modulation). Depending on the setting, this algorithm can be used to create kick, tom, or metallic percussion sounds; higher values for frequency and modulation level produce noisier sounds for hi-hats or shaker sounds. Since the oscillator frequency can be set exactly according to the musical pitch, this model can be used to produce bass lines or melodies.

Physical Modelling: This is a simple physical model of an "abstract" percussion instrument. A fed-back network of delays is oscillated by an impulse of filtered white noise (exciter). Depending on the setting of the exciter, the size of the model (surface), and the damping, a wide spectrum of natural sounding percussion instruments like cymbals, claves, gongs, or triangles can be created.

BeatBox – effects section

Each of BeatBox 2's drum instruments includes an effects unit which is fed in after the actual sound synthesis and editing. Each of these effects units includes a series of high-quality algorithms to add "audio polish" or to place the sound in a production-typical context.

1. **FX on/off**
2. **Parameter controller**: An effects module includes four adjustable parameters. The fourth parameter, "Mix", is always available and the remainder possess a function dependent upon the selected algorithm.
3. **Automation**: This selects the controller's parameter for the automation.
4. **Effects algorithm**: Click the arrow to select an effects algorithm.

The available **effects algorithms** are described in the following:

Mono delay (tempo sync/millisec.)

Simple, monophonic delay effect.

Parameters

Time: Delay time controlled by musical measure (sync) or free

Feedback: Repetitions

Damping: High damping of the repetition

Stereo delay (tempo sync / millisec.)

Stereophonic repetition, separately adjustable per side

Parameters

Left/right: Delay times, synced or free

Feedback: Compared to mono delay, no repetition takes place in the middle position of the feedback controller. When turned to the left, the type of delay is the so-called "ping-pong" variety, i.e. the signal is sent alternately so that it jumps from one channel to the other. When turned to the right, the delay effect is "dual mono", in which case left and right sides function as two independent time-delay units.

Chorus

Produces a typical "floating/shimmering sound" by modulated detuning of a signal to "thicken up" its sound or spread it across the stereo field. Detuning is achieved via a short delay, the length of which can be varied by the modulation. This produces the so-called "Doppler" effect.

Parameters

Time: Delay time in milliseconds. This may be understood as the "base" modulation that is stretched or compressed by the modulator.

Rate: Modulation speed

Depth: Modulation amplitude. Low values modulate only a little; higher values create a clear vibrato.

Flanger

Algorithmically similar to chorus, but different in that the delay time is significantly lower and delay works with repetitions (feedback). A flanger sounds more "cutting" and up-front than a chorus.

Parameters

Rate: Modulation speed

Feedback: Delay feedback

Depth: Modulation of amplitude

Phaser

A modulation effect just like chorus & flanger, but in this case no detuning takes place. Filter components periodically alter the signal's "phase response" (principle of the "phase shifter"). Characteristic notches are produced in the frequency spectrum response to create so-called "comb filter effects". The phaser effect is suitable for pads and "psychedelic" sounds.

Parameters

Rate: Modulation speed

Feedback: Feedback of filter steps

Depth: Modulation amplitude

Room reverb/hall reverb

In the case of reverb, there are two realistic simulations of natural reverberation. Sounds receive "atmosphere" to appear lively and "authentic". Room reverb simulates a small space with high echo density. Hall reverb mimics the typical reverb of large concert halls.

Since natural spaces never sound "static" (air molecules are constantly moving and the reflection process is quite complex), both algorithms include a modulation parameter which varies the delay time of individual echoes and thereby affects the liveliness of the reverb impression depending on strength.

Parameters

Decay: Reverberation length

Damp: Damping of highs to simulate absorption via air, wall materials, and objects

Mod: Modulation strength

Lo-fi

This algorithm gives the sound a little bit of "grit", or a certain measure of signal destruction depending on its setting. An ideal partner for creative sound design. The simulation of early digital synthesizes or samplers is equally possible, since their AD/DA converters were anything but "true" in the character of their sound. The sample rate from the output of the lo-fi effect can be continuously reduced and a generous measure of "aliasing" distortion may be produced alongside the unavoidable loss of highs resulting from "down sampling". Bit resolution is clearly changeable, too.

Parameters

Rate: Sample rate

Crush: Number of bits

Low-pass: Low-pass filter at the output to smooth out induced noise

Distortion

This overdrive effect works similarly to common guitarist effect pedals. Everything is possible, from light, bluesy signal saturation to hard "metal shred boards". In this case, a dual-band EQ works on the in and output signals and to provide a rich palette of sounds.

Parameters

Drive: Controls the internal level and the overdrive

Low: Bass portion

High: High portion

Analog filter 12/24 dB

An additional filter may be applied across the entire drum sound.

Parameters

Cutoff/resonance: Filter frequency and resonance

Type: Filter type

Drive: Filter saturation

Vintage compressor

A compressor with a special "analog" controlling behavior for more pumping bass drums.

Parameters

Input: Input level

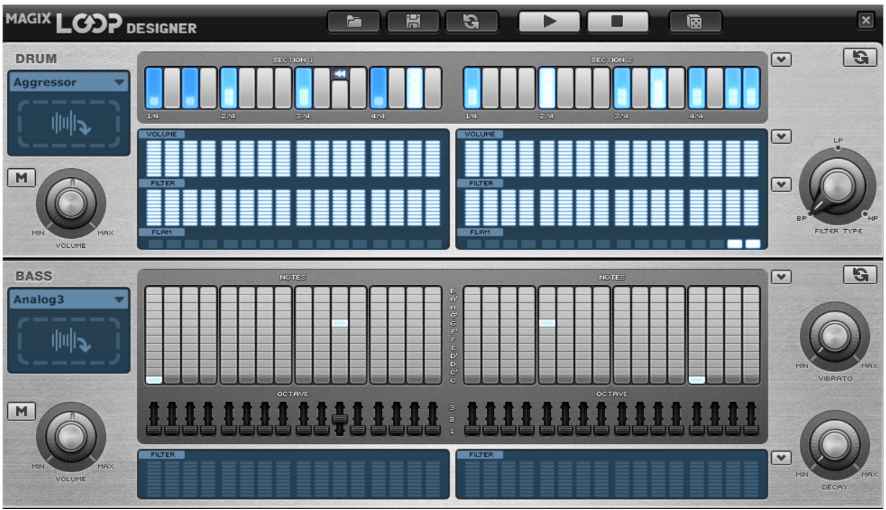
Attack/release: Time constants for compression.

Ratio: Compression ratio.

Loop Designer

The loop designer unites both stellar design elements of drum'n'bass style in a "device": turned-up beats and rumbling bass lines. With the Loop Designer you can create authentic drum'n'bass sounds without any specialized knowledge.

Loop Designer overview



The top half of the synthesizer controls the rhythm section, the lower half - the bass section. You can mute both sections using the M symbol on the left border. This way you can, for example, turn off the bass section in order to take only the drum section into the project. Only the drum section will then be added to the mixdown file when mixing down the project. Next to this are the volume controls, which control the volume of both sections.

You can preview and stop your drum'n'bass creations using the "play" and "stop" buttons.

Additional buttons:

Loads a saved pattern with all settings.

Saves a pattern.

Undo for all settings made in all sections.

Makes random settings changes in both sections. You can change settings later as you wish.

Drum section (top half)

Here complex, authentic-sounding Jungle-Breakbeats can be put together easily. In professional recording studios Jungle-breakbeats are made by chopping up arbitrary drum loops and putting them back together in a different order. These laborious work steps are significantly simplified by Loop Designer.

The new sequence is determined in the top row, the so-called "step" row. The grey cells represent the individual sections ("counts") into which the loop is separated.

Pick a different note or playback by left-clicking a grey cell. Various notes will be represented with a light bar which "grows" toward the top with each mouse click. This way, each time you click on the grey cell, the bar expands by one step.

1 of 4: Play Drum Loop starting from the first note.

2 of 4: Play Drum Loop starting from the second note.

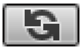
3 of 4: Play Drum Loop starting from the third note.

Full bar: Play Drum Loop starting from the fourth note.

Reverse icon: Play backwards from this position

Stop icon: Stop playback

The right mouse key deletes settings of a step bar and the Drum Loop is played in its original order.



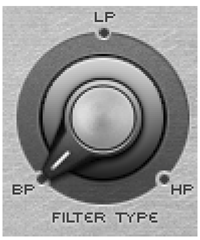
Pressing the double-arrow button generates a random step sequence. You can change this rhythm as you like.

Clicking into the blue field in the left part of the drum section opens a pop-up menu where you can pick the sound of the Drum Loop. When you select another drum loop it will load and be played in the way you programmed it.

Tip: You can also send a loop from Soundpool or a wave file to Loop Designer. To do so, select the desired loop or file and drag it into the field while holding down the mouse key ("drag & drop").

In the field under the "steps" row the sound of the loops is defined. The intensity of the settings is determined with a control similar to a peakmeter. The higher the bar, the stronger the influence on the loop is. With the help of the left mouse key, values can be smoothly adjusted. "Volume" controls the loudness (full = loud, empty = quiet), "filter" the filter strength (full = clear, empty = muffled). In the "Flam" row you have the option to make the note repeat itself twice in quick succession. This is how to program rolls and fill ins.

Using the top arrow buttons at the right edge you can load pre-defined pattern settings. The originally set loop will not be changed. The two lower arrow buttons offer presets in the form of standard curves for "volume" and "filter".



Filter type: selects the type of the filter sound: "BP" stands for "bandpass", "LP" for "low pass" and "HP" for "high pass".

Bass section (lower pane)

The bass section produces suitable bass lines.

- The first row of the "Notes" row, determines the order of the sounds i.e. the series of notes. Select a cell with the left mouse click, where the lowermost represents the lowest note and the topmost - the highest. A right mouse click deletes a cell.

- In the "Octave" row you can specify the octave of the bass. The slide control positions represent the various octave values. If the controller is down, a deep sound is produced, if it is in the top position, you will get a high tone. If the controller is in the middle, the pitch will be balanced.

Just like in the drum section, here you will also find arrow buttons on the right side for opening pre-defined patterns and and a double-arrow button for setting arrows. The filter can be set analogously to the drum section. The arrow button next to the filter area select presets in the form of standard curves.

You can determine the bass sound in the blue selection field at the left side. You can drag loops and WAV sounds into the field using drag & drop just like in the drum section.

Additionally, there are two slide controllers for sound changes to the right:

Vibrato: makes the bass tone's pitch change. When the control is to the far right, the sound vibrates more, if it's to the far left pitch isn't changed at all.

Decay: determines how long should it take to for the sound to subside. Set to the far right the sound is very fast (about 1 / 4 of a second), on the far left the sound lasts longer.

Robota

Robota is a four-voice drum machine with a virtual analog generator and a sample-based synthesizer. Virtual-analog sound generator means that the sounds are synthesized in realtime, i.e generated by a synthesizer. This gives you all of the typical analog sounds of classic drum machines like the Roland TR-808, TR-909 or more modern models like the Korg Electribe or the Jomox X-Base.

In the sample sound generator, samples of drum sounds or other recordings are loaded and used as a basis for creating the sound. After selecting the basic sound generator, the sound characteristics of each of the four voices can be fine-tuned using various controllers.

Robota plays back through a step sequencer with light chaser programming. The four bars run as a loop in semiquavers (16ths) or two bars in demisemiquaver (32nds) steps. You can set the playback position from any beat position by clicking with the mouse. In Event mode, the instruments are arranged on the beat grid. In Snapshot mode you can adjust the various sound settings of the instruments.

Robota – sound synthesis

The four voices of Robota are built identically. The sound synthesis in Robota is quite complicated because each voice has to generate a large variety of drum sounds, ranging from a hissing hi-hat to heavy bass drum sounds.

The Robota drum synthesizer consists of an oscillator with a selectable waveform (sine, triangle or sawtooth) or a sample. You can also use a noise generator to add some noise to the mix. The oscillator features a pitch envelope and attack/decay controls. It also features frequency and ring modulation. The modulation depth can be controlled by means of an envelope parameter (FM/rng dcy). This is followed by a "Lo-Fi" section consisting of distortion (rectify, diode rectification), bitrate reduction (crush) and sample reduction (down sample).

The intensity of the Lo-Fi effects can also be adjusted by means of an envelope (lo-fi dcy). This is followed by a multimode filter (low-pass/bandpass/high-pass) with a roll-off of either 12dB or 24dB. A comb filter (a delay line named after its spectral view of peaks and dips that resembles a comb) can also be added. You can also modulate the filter frequency by using an envelope. There is also a compressor that can be adjusted for response time and intensity ('compressor' comp resp) and a tube amp simulation, both of which will provide the necessary final sound pressure.

To simplify this a little, when editing a sound the only **Parameters (1)** that can be changed are those which are appropriate for the selected drum sound (Snare, Kick, Hi-Hat etc).

There are four selected variable parameters, which are precisely tailored to the selected preset sound.

Oscillator Waveform (2): The fundamental waveform of the oscillators is selected here. You can choose between a sine, triangle, or sawtooth sample. If "Sample" is selected, you can use the controller to select a sample, i.e. a previously recorded drum sound. You can find these samples in the folder ./MxSynth/Robota/Samples. If you save samples in this folder, they will appear in the selection list.

(3): For each voice the Filter Cut-off, Resonance, Tube, Volume and Panoramaare always adjustable

With Select you can choose the instrument you want to edit in the step sequencer.

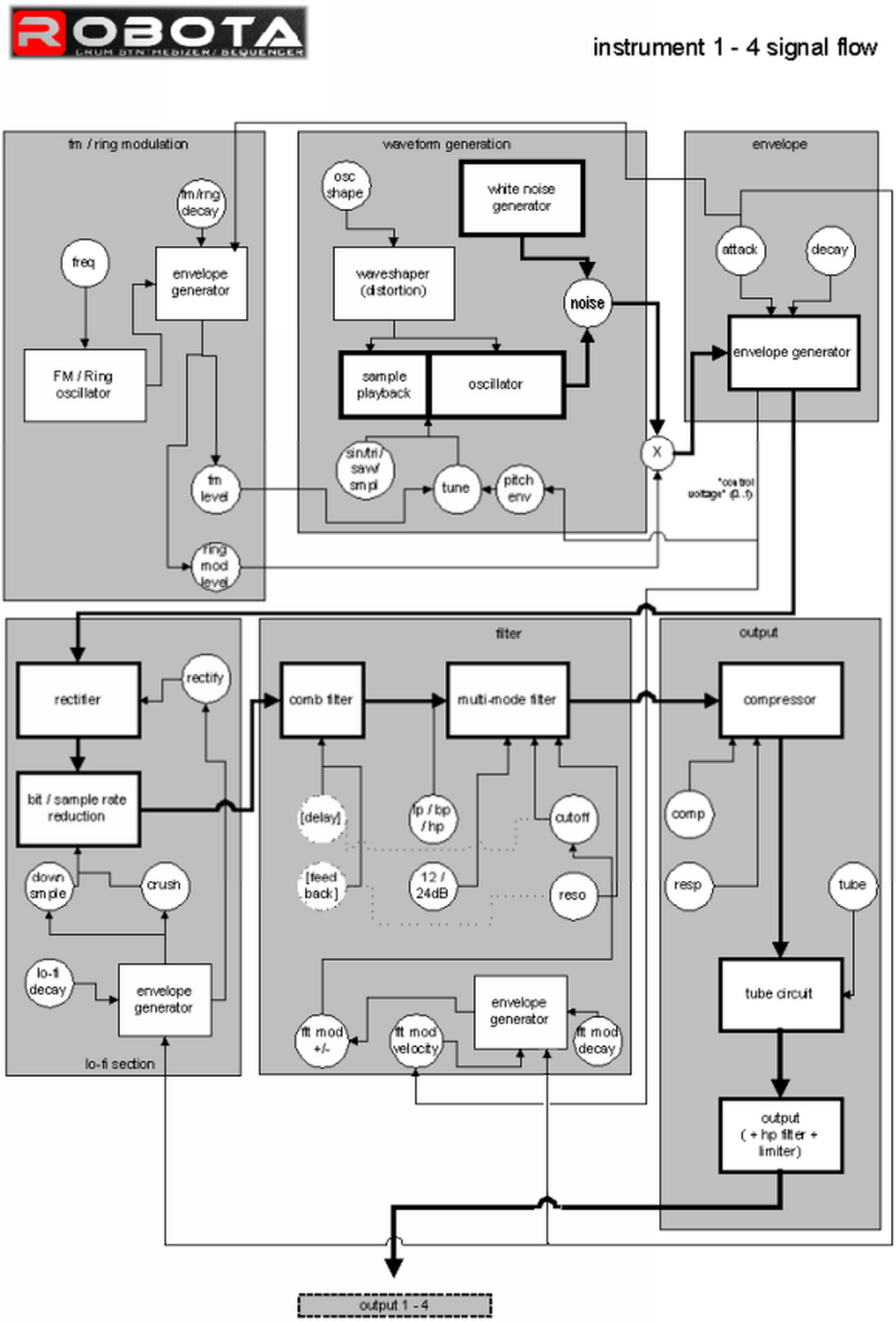
Click on "M" to set the instrument to mute, "S" for solo.

You can preview the instrument by pressing the loudspeaker button.

Master section

With volume you can control the master volume of the Robota. Distortion adds a controllable tube distortion to give the sound more power and make it sound rougher. The peak meter is used to control the output level. If this lights up in the red area, decrease the master volume.

Robota sound synthesis – schematic illustration



Detailed Circuit Diagram of a Robota Voice

Description of all control parameters:

Pitch Envelope (pitch env): Controls the pitch envelope level.

Tune: Tunes the instrument.

Oscillator Shape (osc shape): The shaper adds additional frequency components to the basic sound of the oscillator by artificially reshaping the waveform. A sine wave (shape = 0) can be reshaped up to a square curve (shape = max).

Oscillator Waveform: The fundamental waveform of the oscillators is selected here. You can choose from sine/triangle/sawtooth/sample. If "Sample" is selected, you can use the controller to select a sample, i.e. a previously recorded drum sound.

Noise: Here you can adjust the ratio between the oscillator sound and the noise generator.

Attack: This is the ascent time of the volume envelope curve. The higher the set value, the greater the length of time it takes for the sound to surge in. The attack value also applies to the LoFi and filter envelopes.

Decay: This is the descent time of the volume envelope curve. The higher the set value, the longer it takes for the sound to fade out.

FM/Ring modulation frequency (Fm/rng frq): This is the fundamental frequency of the frequency or ring modulation.

FM Level (fm lvl): At a low frequency FM first adds vibration to the sound, at high frequencies and low levels it creates bell-like sounds, as the level increases metal sounds, and finally noise.

Ring modulation level (rng lvl): Ring modulation creates typical auxiliary frequencies.

FM/Ring modulation decay (Fm/rng dcy): The time constant of the by-product of FM/ring modulation. Only the beginning of the drum sound is affected by the modulation.

Rectify: Distorts the audio signal

Crush: Bitrate reduction. Digital artifacts become audible with higher settings.

Down sample (dwnsmple): Sample rate reduction is useful for creating the "old school" sounds of older digital drum machines. The higher the set value, the more muffled the sound becomes.

Lo-Fi decay (lofi dcy): The time constant of the by-product of the three lo-fi effects for making the sound "dirty". Only the beginning of the drum sound of the lo-fi effects is affected if the decay is low. For instance, this makes the kick of a kick drum sound more interesting.

Filter modes (flt mode): The various modes of the filters:

- Low-pass (LP) - Sound portions above the cut-off frequency are filtered out.
- Bandpass (BP) – Sound portions above and below the cut-off frequency are filtered out.
- High-pass (HP) - All sound portions below the cut-off frequency are filtered out.

The mode is set as a preset and cannot be changed.

Filter frequency (flt freq): The cut-off frequency of the filter.

Filter resonance (flt reso): The sound portions are amplified at the filter cut-off frequency. If the resonance is high, the filter itself can also be used as an oscillator.

Filter-Modulation decay (flt mod dcy): Regulates how much and in which direction the filter envelope curve moves the filter frequency.

Filter-Modulation decay (flt mod dcy): Decay time of the filter envelope. Smaller values with high resonance create a "zapping" sound of the filter, greater values create the typical sweep sound.

Filter modulation velocity (flt mod vel): Specifies how much the filter modulation depth depends on the velocity. If this value is increased, louder beats will generate higher filter curves than quieter ones.

24 dB: The filter can operate with a slope of 12 dB or 24 dB. This mode is set as a preset and cannot be changed.

Comb filter (comb filt on): A comb filter can be activated here. This effect produces a feedback delay with a very resonant sound similar to a plucked string. The delay time and feedback levels are permanently linked to the filter parameters (frequency and resonance). This comb filter is set as a preset and cannot be changed.

Compressor: Controls the compressor strength. This lets you increase the "power" of the drum sound.

Compressor response (comp resp): Controls the compressor time. The lower the value, the faster the compressor readjusts the volume.

Tube: Controls the level of the tube amp simulation. This "saturates" the output signal of the voice and adds warmth to the sound if the settings are moderate. Increasing the settings makes the sound "dirtier".

Volume/Pan: Controls the volume and panorama position of the drum instruments.

Robota – sequencer

Classic light chaser programming, just like what you find in most of the classic drum machines and groove boxes, is used to control the drum patterns. The step sequencer consists of 16 individual step buttons with LEDs which correspond to the sections of a bar (16 or 32 beat, whereby only half a bar is displayed).

A flashing button corresponds to the drum sound at this point in the bar.

Clicking the button turns the step on and another click will turn it off again.

The maximum length of a drum pattern is four beats. The length can be selected in the small scroll bar above the toolbar.

The bar you want to edit can be selected with the corresponding "Edit" button. Use the “Follow” button to select follow mode, i.e. the step display follows the steps of the currently played beat.

„**1>2-4**“ **Auto Draw:** If more than one bar is set as the pattern length, "Auto Copy" mode ensures that the drum notes set in the first bar are automatically placed into the next bars. This also makes it easy to create a continuous beat, even with a loop length of four bars. Notes set in the bars further back are not affected by the "Auto Draw" function, e. g. making faint variations detectable only in the fourth bar.

Programming a new drum pattern:

- Use the slider to select the pattern length.

- Select “Event” mode.

- To make changes during playback, deactivate "Follow" and select the bar you want to edit with the "Edit" button.

- Use the “Select” button to select the instrument to be edited.

- The “Clear bar” button deletes all tracks in the current bar of the selected instrument.

- Turn on the corresponding step buttons. With the „Velocity“ slider you can set the velocity of the beat.

- Repeat this process for the other instruments.

Snapshots

It is also possible to automate the editable sound parameters of a drum sound with so-called “Snapshots”. This enables the sound parameters of an instrument to be saved on the step buttons of the sequencer.

Here's how to automate a drum instrument using snapshots:

- Set edit mode to “Snap”.
- To make changes during playback, deactivate "Follow" and select the bar you want to edit with the "Edit" button.
- Select an instrument and change the sound according to your preferences. You can check the sound of an instrument once playback is stopped by clicking the loudspeaker button or by triggering the sound by MIDI (requires activated monitoring in Samplitude).
- Save the sound as a snapshot to one of the step buttons.
- You can now change the sound of the drum instrument and save the settings to one of the other step buttons.

Note: Changes to parameters are not executed immediately but are blended in to avoid clicks and pops. When you play the pattern the drum sounds will sound different to what you expect if two snapshots with extreme parameter differences are too close to one another.

- You can switch between individual snapshots using the arrow keys when playback is stopped.

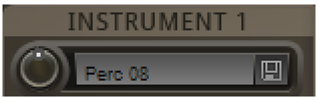
- Now activate the snapshot automation with "on".

Groove-control

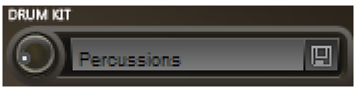
The secret to “groovy” beats is to play back the individual beats delayed or earlier according to certain patterns. For example, shuffle is used for house beats, whereby the even 16ths are delayed.

Robota features Groove Velocity and Groove Swing presets. Groove Velocity presets increase or decrease the original velocity of each step in a bar according to a particular offset for each step. Groove Swing presets apply a time offset to each step whereby the beat is played slightly earlier or later than the original step. The result is a considerably more lively-sounding drum sequence. The strength of each effect can be selected with the "**%**" controller.

Robota – setups, drum kits, presets and patterns

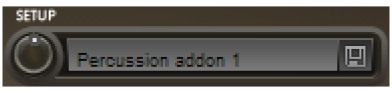


An individual drum sound is saved in a Preset.



The presets of the four voices can be saved together as a Drum kit

All note information and snapshots are saved in a pattern.



The combination of everything (instruments + pattern + further settings) is a Setup.

Load/Save: You can select presets, drum kits, patterns and setups using the knobs beside the display fields. To save click on  the "Save" button. At this point you can enter a new name in the data entry field. Press "Enter" to save.

Note: A drum kit only saves the names of the presets and not the actual parameters. This means that if you have prepared your own drum sounds by changing existing presets and would like to save these as a drum kit, you first have to save the drum sounds as new presets and then the new drum kit. The same is true for setups that only contain the pattern and drum kit name.

It is always best to save in the following order: Preset > Drum kit > Pattern > Setup. This only applies for creating your own "Templates". If you save your arrangement, the complete current status of Robota (synthesizer + sequencer) will be saved with it.

MAGIX Synths

The following software synthesizers and samplers based on VST plug-in technology are also integrated in Samplitude:

- **Independence**: A sampler with 12 GB of content for Samplitude Pro X6 and 70 GB Content for Samplitude Pro X6 suite or Sequoia.
- **Revolta 2:** A polyphonic analog synthesizer based on subtractive sound synthesis with frequency modulation. Sounds in Revolta 2 don't use samples as a foundation - they are calculated in real time from your computer processor. Revolta 2 is particularly suited for electronic music, with lead, sequencer and pad sounds as well as bass and effects.
- **Vita**: A sampler with incredibly realistic-sounding, "classical" instrumental sounds like different guitars (Power Chords, clean electric guitar, acoustic guitar, bass guitar), different pianos, percussion, strings, brass, woodwinds (each as an individual set & as an ensemble set), and much more .
- **Vita Solo Instruments** is a collection of different sample players based on Vita sampler technology and with customized interfaces for each of the instruments.

DN-e1

The DN-e1 is a virtual analog synthesizer that is suitable for all conceivable styles and application areas. It works in a subtractive way, i. e. first a basic sound is selected that is then filtered with the aid of a filter curve.

The DN-e1 is played with a MIDI keyboard or with the aid of MIDI objects. You can use the keyboard in the program to set the sounds.

Sound selection

Select the sounds and sound configurations at the top.

Bank: Here you can switch between three banks with various complete configurations.

Category: Here you can select a sound category.

Rndm (Random): Here you can activate a random selection of the parameter settings in order to experiment with the sound.

Patches/Name: Here you can select a sound that will then be modulated.

Output

The end of the signal chain is edited in this area.

Volume: Sets the total volume.

Voices: Controls the number of voices generated (polyphony).

Glide: Controls the glide function. You can access sliding pitch transitions between the individual notes.

Unisono: Switches to monophonic, but generates a number of slightly varied voices for "Thickening" the sound.

Filter en.

In this area the filter curve used to filter the output sound is modulated.

Attack: Sets the time duration that the filter curve requires in order to reach its maximum.

Decay: Sets the time duration that the filter curve requires in order to go from its maximum to the sustain level.

Sustain: Here you can set the degree of filtering that should take place after the decay phase. This filtering remains the same until the key on the keyboard is released; in contrast to the other three parameters, it does not also control a time duration, but a specific level.

Release: Sets the time duration which the filter curve requires in order to go from the sustain level to the zero point after the key is released.

Reverb

An additional reverb effect can be set here.

Type: Sets the sound coloration of the reverb effect.

Pre Del: Sets the time that passes between the direct signal and the arrival of the early reflections. The reverberation time comes only after this time span.

Damp: The corner frequency at which a damping of the highs should be implemented for each delay is defined here. This is useful, for example, for making the delays reverberate more naturally or for creating special effects (reggae/dub-style effects).

Decay: Sets the complete reverb time.

Low Cut: Sets the filter frequency of a high-pass filter. All signal components below this frequency will be filtered out.

Amount: Here you can set the mixing ratio between the effect and the pure sound, i.e. the original sound without any effect applied.

Delay

An additional echo effect can be set here.

Type: Different types of echo can be selected here: normal echo, ping-pong echo (where the sound swings through the stereo panorama) and various other forms.

Color: Sets the sound coloration of the echo.

Feedback: Sets the number of echo repeats.

L Rate: Sets the time duration for individual echoes for the left channel.

R Rate: Sets the time duration for individual echoes for the right channel.

Amount: Sets the mixing ratio between the effect and the pure sound, that is to say, the original sound without any effect applied.

Independence

The Independence sampler gives you access to hundreds of music-oriented software functions. The intuitive user interface and file management system, the ultra fast streaming integration, multi-core processor support and the Auto-RAM-Cleaner makes it possible to store and use a huge number of instruments in seconds.

Detailed information about "Independence" can be found in the PDF document.

Revolta 2

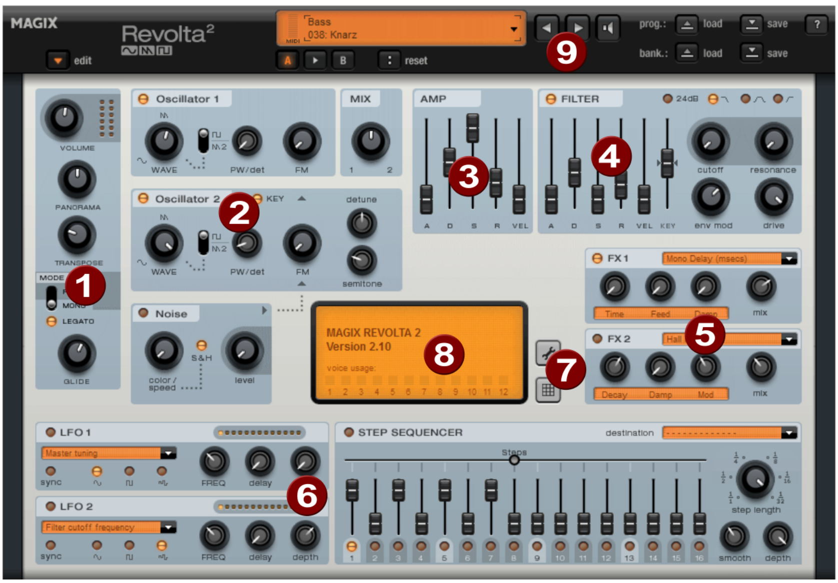
Revolta 2 is polyphonic and may be played with up to 12 voices. It includes a series of additional features, like an additional noise generator, a step sequencer, and an extra flexible modulation matrix. Its effects section of 9 different effects and presets, designed by a famous designer, make it a full-fledged synthesizer for all kinds of lead, sequence, and pad sounds.

Revolta 2 has a whole array of presets. The sounds have been created by professional sound engineers and demonstrate the huge potential of this instrument right from the start. But first of all, we encourage you to try out the various control functions and to experiment as much as you like. The sky's the limit to your creativity.

Revolta 2 – the interface

The Revolta 2 interface may be displayed in two sizes. In "Rack" mode, only the elements necessary for preset loading are visible:

Clicking the "Edit" button opens the entire user interface.



1. Main parameter: The volume, panorama position, pitch characteristics ("transpose") and the play modes (POLY, MONO, LEGATO) may be set here. GLIDE controls the portamento time.

2. Oscillator section: Two oscillators are available with a smoothly adjustable curve form and a noise generator. Both oscillators may be tuned to each other and used to modulate frequencies.

3. AMP: This is the volume envelope. This influences the temporal progression of a track's volume. A(ttack) stands for the volume increase at the start, D(ecay) for the length of time the decrease in volume takes on a section set with S(ustain) at the maximum volume. R(elease) is the length of time it takes for the sound to ring out. VEL specifies how much the envelope curve depends on the velocity.

4. FILTER: Switch on a filter to influence the sound here. FILTER TYPE selects the filter type. "Cut-off" regulates the filter frequency; "Resonance" controls the level of amplification of the filter frequency. VEL" indicates how much the velocity influences the filter frequency, "KEY" changes the filter frequency depending on the note pitch ("Keytracking"). The filter envelope (ADSR slider) influences the filter frequency depending on the time. "Env mod" controls the strength of the filter envelope curve, and "drive" may be used to overmodulate the filter.

5. FX1/FX2: Mix in 2 different effects out of a total of 9 available effects.

6. LFO1/LFO2/STEPSEQUENCER: Two LFOs and the step sequencer may be used to modulate single parameters of Revolta 2.

7. Options and modulations matrix: The two buttons open Revolta's options page for general and preset-specific settings and modulation matrix. In the modulation matrix, modulation sources are connected with modulation targets. Simple modulations like the oscillator, where the pitch will be modulated via an LFO (vibrato), may be set quicker directly on the interface. Much more complex modulations are possible in the matrix, because the matrix offers more modulation sources (e.g. MIDI controller, oscillators) and the modulation source may influence more targets.

8. VALUE DISPLAY: The value display shows the exact value of the parameter that has been modified. You may also use this to find out the load of the twelve voices.

9. Preset section: Select different Revolta presets here. Every sound may be previewed, and an A-B comparison between two sounds is also possible (e.g. an edited and the original sound).

Vita Solo Instruments

The Vita Solo Instruments are contained in a sample player with customized interfaces for each of the instruments.

The basic controls are identical for all instruments.

One click on the arrow symbol opens a fold-out menu where you can determine the general sound of the instrument. If "ECO" appears in the description, this refers to especially performance-improving settings which may not sound so "smooth". You can also save the settings you changed in order to add them to your favorites lists for later use.

You can control the overall loudness of the instrument.

You can turn the instrument keyboard on or off with this controller.

Vita

The MAGIX Vita synthesizer specializes in the realistic reproduction of instruments. It works using sampling technology which combines short recordings (samples) of real instruments of various pitch, playing techniques and volume, which are then combined and reproduced.

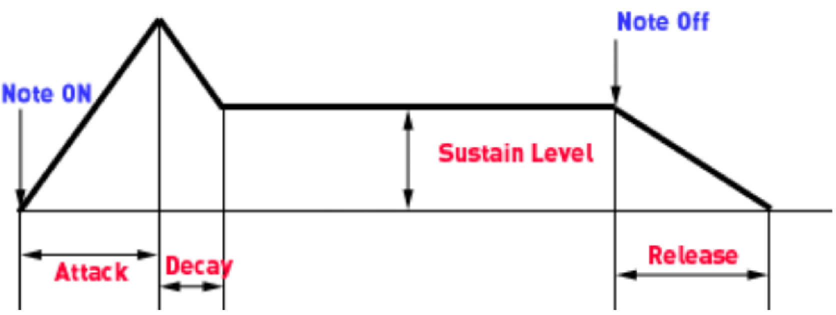
The Vita synthesizer is controlled using MIDI objects.

Vita – the interface

1. Layer selection/peak meter: The Vita sounds, also known as layers, may be selected here using the arrows. Right clicking on the display opens the layer menu.

2. Main parameter: Set the volume, panorama position, pitch characteristics ("transpose"), and the fundamental frequency ("master tune") here.

3. AMP: This is the volume envelope. This influences the temporal progression of a track's volume. A(ttack) stands for the volume increase at the start, D(ecay) for the length of time the decrease in volume takes on a section set with S(ustain) at the maximum volume. R(elease) is the length of time it takes for the sound to ring out.



4. FILTER: Switch on a filter to influence the sound here. FILTER TYPE selects the filter type. "Cut-off" regulates the filter frequency; "Resonance" controls the strength of the amplification of the filter frequency. "Velocity" specifies how strongly the velocity should influence the filter frequency, while volume can be balanced using the "Gain" controller. The filter envelope (ADSR slider) influences the filter frequency depending on the time.

5. DELAY: Switches on an echo effect; "Time" controls the delay time, "Level" controls the strength of the echo sound.

6. REVERB: Switches on a reverb effect, "Time" controls the delay time, and "Level" controls the strength of the echo sound.

7. VALUE DISPLAY: The value display shows the exact value of the parameter that has been modified.

8. DYNAMIC RANGE: Usually, the relationship between the created volume and MIDI velocity is proportional. Since some MIDI keyboards produce a velocity that is too hard for loud sounds or vice versa, this behavior may be compensated with "MIDI input curve". "Dynamic" and "dynamic curve" influence the dynamics of the sound, i.e. the relationship between the quietest and loudest sounds.

9. Voices: Control the number of voices played simultaneously here. If notes are no longer played during fast passages, increase the amount of voices here (performance will not be as good).

10. Keyboard: A preview of the Vita sounds. This only works during playback or recording.

Vita – MIDI specifications

Vita parameters	Control Change #	GeneralMIDI
ModWheel	1	ModWheel
Volume	7	Volume
PAN	10	PAN
Mastertune	14	Mastertune
Sustain	64	Sustain
Cutoff	71	Filter Cutoff
Amp Release	72	Release Time
Amp Attack	73	Attack Time
Resonance	74	Resonance
Amp Decay	75	Decay Time
Amp Sustain	80	Button 1
Filter Velocity	81	Button 2
Filter Gain	82	Button 3
Reverb Level	91	Reverb
Reverb Size	92	Tremolo
Delay Level	93	Chorus
Delay time	94	Delay/Vari
Filter Attack	102	-
Filter Decay	103	-
Filter Sustain	104	-
dynamics	108	-
dynamics curve	109	-
MIDI input curve	110	-

Vita MIDI Event Types

Pitchbend	x
NoteOn	x
NoteOff	x
ControlChange	x
Aftertouch	0

Plug-ins at Track, Object, and Master Level

MAGIX Plug-ins > Analogue Modelling Suite: AM-Munition, AM-Phibia, AM-Pulse, AM-Track **(Samplitude Pro X6 Suite)**.

AM-munition is an extremely versatile, dynamic tool for editing groups or signal sums, especially in the domain of mastering. It has separate units like compression, filtering, side-chain, limiter and clipper. All modules and parameters are optimized to perform the fundamental function without any compromises: Effective enrichment of the program material without causing bothersome artifacts, a high reachable volume and an 'analogue' behavior with an individual sound signature.

AM-phibia is a tube amplifier/channel strip. It combines an optical compressor with a pre and post filter unit. The filter presets offer appropriate settings for various types of input signal. When interacting with the compressor section, am-phibia may be used as a vocals pre-amp, tube guitar amp, or simply for creating a warm sound.

AM-pulse is a "transient modeler", a creative tool for editing envelope and sustain processes on percussive or dynamic signals.

AM-track is a combination of an analog compressor and a tape simulator in a single device. This is used primarily for so-called "tracking", i.e. editing individual channel strips or subgroup signals.

MAGIX Plug-ins > Vintage Effects Suite: CORVEX, ECOX, FILTOX

This suite enhances the repertoire of Samplitude with a chorus/flanger, delay and filter plug-in. **CORVEX**, **ECOX**, and **FILTOX** are each based on the same basic handling principle: as required, a modulator (LFO) controls almost all knobs accessible on the "front panel".

MAGIX plug-ins -> VariVerb Pro

VariVerb Pro is a classic reverb unit which performs classic and modern algorithm-based reverb generation without accessing impulse responses. The effect includes several rooms, halls, hall plates and non-linear algorithms that can be edited in two different modes in many ways.

Note: You can open the MAGIX plug-ins in the plug-in section of the mixer, track editor, track, or at the object level through "Effects". Detailed information about MAGIX plug-ins can be found in the help file "Effects -> MAGIX Plug-ins (view page 849)".

VST/Rewire: All plug-ins installed on your system are written into the Windows

Registry and are available through the effects slot or by going to "Object -> Object

Effects > Object Plugins. At track or master level, you can access plug-ins by clicking

on the "FX" button.

VST FX: This is a list of all VST effect plug-ins included in the VST plug-ins folder. The path for VST plug-ins can be determined via "File > Program preferences > System/Audio (view page 615)", shortcut "Y".

Vandal

(VANDAL in Samplitude Pro X6 Suite / VANDAL SE in Samplitude Pro X6)

The main concept of the VANDAL guitar amp (view page 919) consists of virtual switching that models three different preamps and two power amps. This can be achieved by switching the mode functionality on an easy, single front view.

essentialFX

The essentialFX Plug-in Suite offers high-quality audio tools for a wide range of tasks. Each plug-in has a number of important qualities for producing quick and accurate results:

- Low-resource DSP functions despite top quality algorithms
- Low space requirements for interfaces
- Easy operation thanks to reduction of function elements featuring only the most important parameters
- Consistent operation of all interfaces
- Simple, straightforward preset management with the tried-and-true file/folder basis; simple upgrades for users
- Color coding: Each plug-in has a color code in the top part of the window for quick identification; modulation effects = blue, dynamic plug-ins = orange

The following essentialFX are available: eFX_Reverb (view page 853), eFX_StereoDelay (view page 855), eFX_ChorusFlanger (view page 851), eFX_Phaser (view page 852), eFX_Compressor (view page 857), eFX_Limiter (view page 861), eFX_Gate (view page 859), eFX_DeEsser (view page 864), eFX_VocalStrip (view page 866), eFX_TubeStage (view page 862), eFX_TremoloPan (view page 865).

Melodyne integration

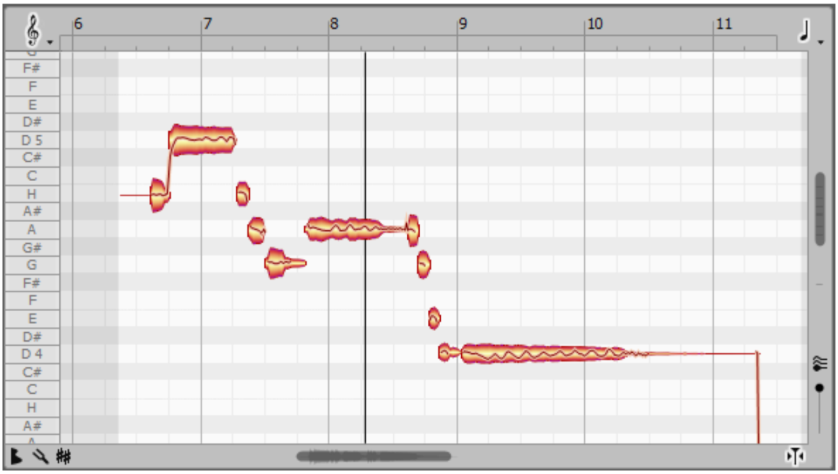
Melodyne is a software that lets you make extremely detailed edits of pitch, timing and spectrum of audio material, as well as change the melody of individual instruments within a complete mixes. It was possible to use Melodyne in older versions of Samplitude as a VST plug-in, and starting from Samplitude X5, a special program interface from Melodyne (ARA = Audio Random Access) had been implemented, letting Melodyne be seamlessly integrated (starting from version 4) into Samplitude interface.

Melodyne essential, the basic version of Melodyne, is included with Samplitude. You can find more information about Melodyne capabilities and its various editions on the manufacturer's website from Celemony http://www.celemony.com/de/melodyne/editions-and-technical-matters.

Edit audio objects with Melodyne

To edit one or multiple audio object with Melodyne, select them, right click and select "Edit audio file in Melodyne" from the context menu.

Melodyne will be loaded for these objects as an object effect, the objects will be analyzed and the Melodyne screen will be opened. There you will see the last select object as a typical Melodyne "blob".



The blob view is a mix of piano roll and waveform display.

You can copy, move, lengthen, shorten and play back all notes here. The scope of possible edits depends on the Melodyne version and spans pitch and timing changes up to detailed intonation control in polyphone material and spectral editing of multiple tracks.

Tip: To be able to use Melodyne to its full capacity, we recommend you watch the video tutorials on the Celemony website http://helpcenter.celemony.com/hc-2/!

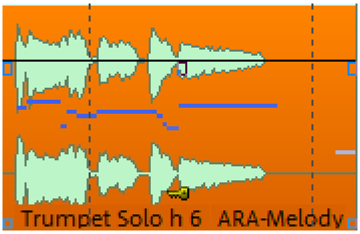
If you select another object, the display in the Melodyne window will adjust accordingly.

You can let the "blobs" representing the other objects appear for reference purposes by activating this symbol to the left in the Melodyne window.

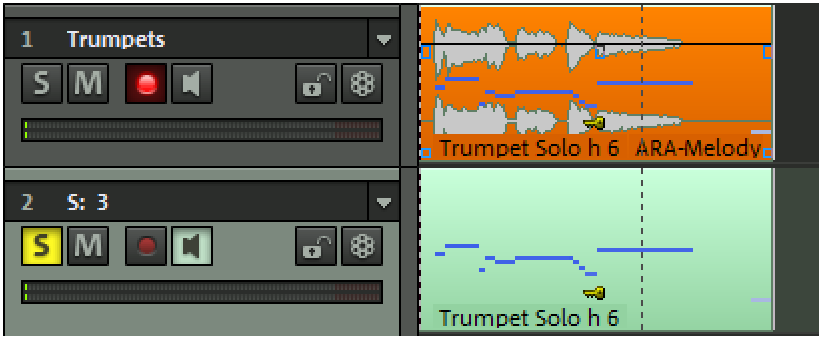
If the objects originate from different tracks, these will be analogously created in Melodyne (only Melodyne studio), with the track names taken from Samplitude.

Audio to MIDI

If an audio object has been analyzed or edited using Melodyne, you will also see the pitch on the audio object as a piano roll, similar to how MIDI objects in VIP are represented.



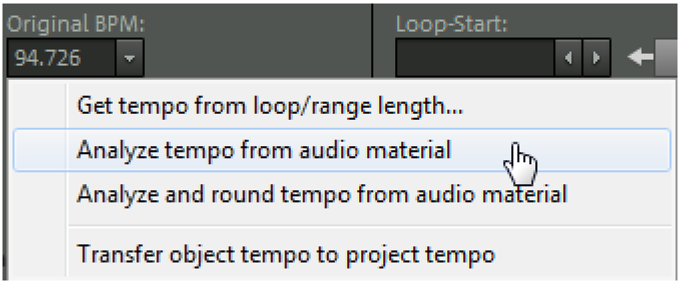
Selecting "Audio to MIDI..." from the object context menu or the "Object" menu, you can create a new MIDI object.



The MIDI object will be added to the track directly below the audio object. If there are already objects at this time position, on this track the MIDI object is inserted to a newly created track instead.

Determine object tempo

If Melodyne is installed, it can be used to determine object tempo. You can learn more in the Object Editor section, Pitchshifting/Timestretching (view page 155)



Notes

- The VST 3 version of Melodyne is required to work with Samplitude via ARA.
- Melodyne integration via ARA works only if you set Melodyne as an object effect, which is easiest set in the object context menu. You can also load Melodyne from the object editor or the Effects menu.
- If Melodyne is loaded in an audio object, right-clicking on the object will open the Melodyne Editor, not the Object Editor. Object Editor can be accessed via the context menu, the Object menu or the Ctrl + O keys.
- Melodyne can't be used simultaneously with Samplitude tempo and pitch functions (Elastic Audio, Timestretching/Pitchshifting, musical tempo adjustment).
- If an audio object is split, Melodyne edits will be applied to both objects, just like if this object is duplicated with the command or using Ctrl + mouse drag. Melodyne edits will not be transferred if you copy an object to the clipboard and then paste it!
- Object editor functions reverse and loop aren't allowed.

Surround Sound

Samplitude offers several possibilities for mixing audio material in various multi-channel formats - from LCR (Left-Center-Right), to Quad (Left-Right-Left Rear-Right Rear) on through to various 5 and 6 channel formats.

A prerequisite for this is a mixer configured with a Surround Master corresponding to the target format. Some target formats are 5.0 ITU, 5.1 DD and 6.0 DTS-ES.

In the „Surround Settings“ you can assign Surround Master channels to available playback devices. The audio material of the tracks in an arrangement that are routed to a Surround master can be distributed across the available surround outputs in the Surround Editor. In the mixer, the panorama knob in these tracks is replaced by a Surround panorama display.

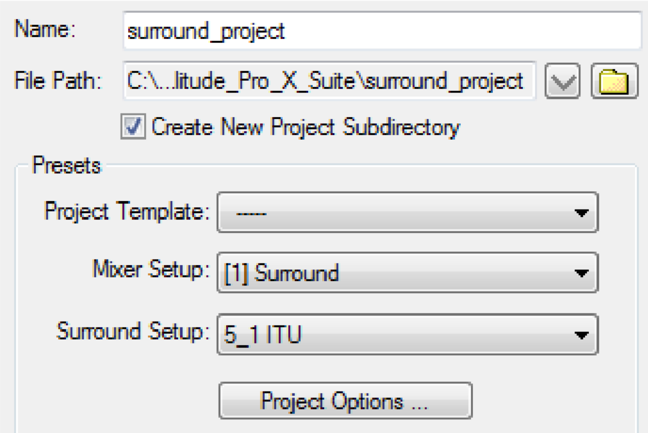
You can also route individual objects directly to the surround master and distribute them in the surround panorama, regardless of whether the track on which they are placed is routed to a stereo or surround master.

The Surround Editor provides four different modes for routing the signal into the Surround panorama. Various surround effects are provided in the mixer for mixing in multi-channel formats.

You can create any number of additional master buses, e.g. in addition to a 4.0 mix you can create a 5.1 mix in the same project. A master bus can also be routed to any other surround or stereo master, while the required upmix or downmix is performed.

Create new Surround Project

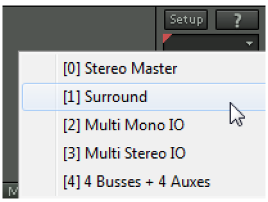
In the "Setup for new Project (VIP)" dialog, select the option "Surround" in the "Mixer Setup". Select the desired format under "Surround Setup" (e. g. 5.1 ITU).



In addition to the audio track channels, the mixer in the new VIP now also contains the Surround Master in the selected format (e. g. 5.1 ITU L, R, C, LFE, Ls, Rs). All mixer channels for the corresponding project tracks are automatically routed to the Surround Master. The mixer's Stereo Master will be hidden initially, but it can be displayed by clicking the "hide Master" button in the mixer. Routing of the individual mixer channels to both the Surround Master and the Stereo Master is possible.

Converting an Existing Stereo VIP into a Surround Format

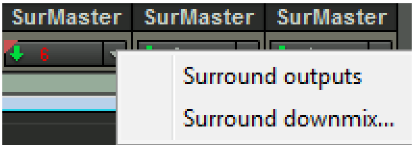
In the mixer of an existing VIP click on the mixer "Preset" selection button at the bottom right corner under the "Setup" and select "Surround". This opens the „Surround Setup“ (view page 298) dialog where the you can select the surround format (e. g. 5.1 ITU).



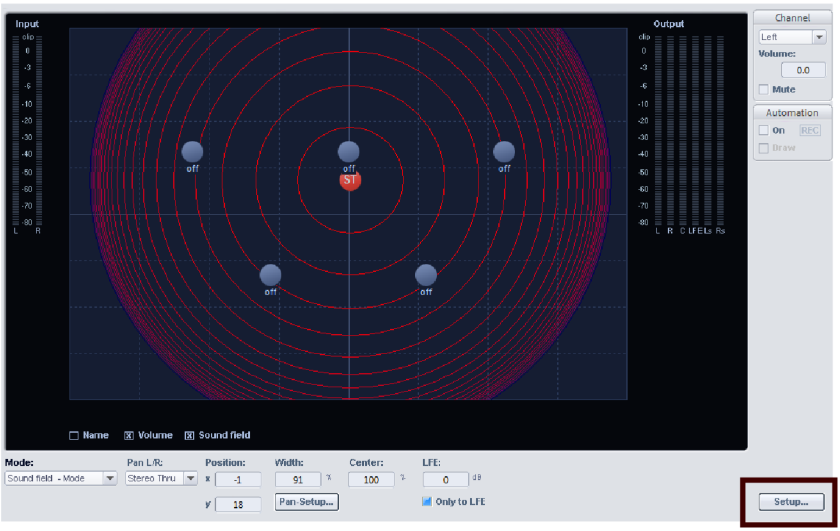
Surround Settings

Setting the Physical Outputs of the Surround Buses

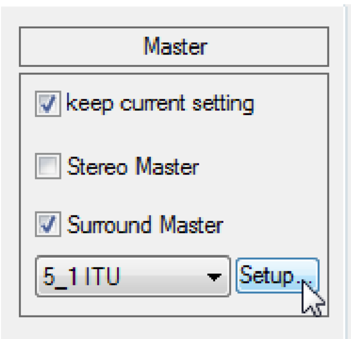
The "Surround Setup" dialog window opens when you click on the outputs of the of the Surround Master in the mixer and select "Surround outputs..."



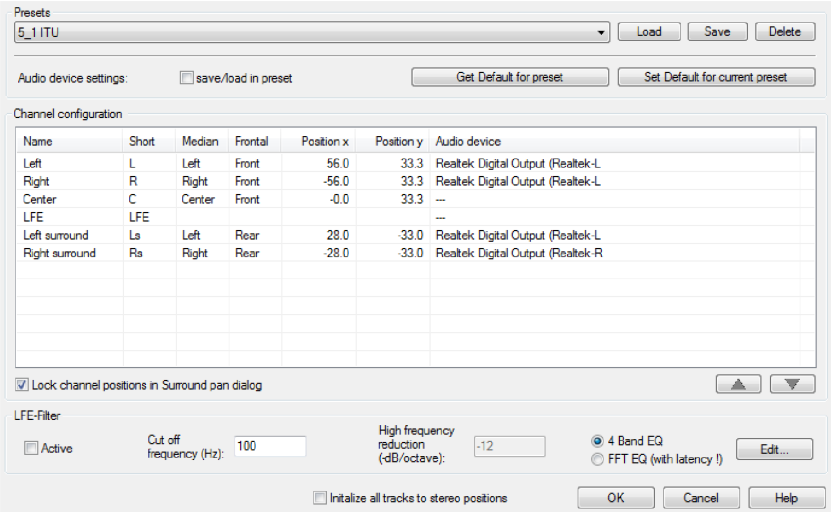
or when the "Setup..." button in the „Surround Editor“ is clicked. (view page 303)



It can also be opened by clicking the "Setup..." button in the master section of the „Mixer Setup“ (view page 598).



The surround format for mixing is set in the "Surround Setup" window. Here you can open various saved presets (e. g. 5.1 ITU, DD, DTS, LCR, LRS, Quad...) or edit and save your own settings. The filter settings for the LFE (Low Frequency Effect) channel are also set here.



Presets

Previously added Surround formats may be accessed and new formats may be saved here. A preset features the Surround buses, their name as an abbreviation, the sequence, and the position coordinates of the speaker. When loading a preset, the playback device settings will feature the default values set for the preset.

Loading Standards for Presets

Clicking on the buttons loads the default configuration of the playback devices belonging to the currently active preset.

Setting the Default for the Current Preset

Pressing this button saves the current assignment of the Surround buses to the playback devices as a configuration of the currently active preset. This configuration is then always loaded when the preset is used.

Loading & Saving Audio Device Settings with Presets

Activating this option saves the current playback device configuration during playback, independent of the default configuration in the preset.

Channel Configuration

Name Determines the name of the individual surround buses. The abbreviation in the following column is generated from this name. The first letters of each word (or capitalized letters) are used for this abbreviation.

Left = L

Right = R

Center = C

Left surround = Ls

Right surround = Rs

Low Frequency Effect = LFE

These abbreviations are used to label the corresponding channels in the mixer and in the Surround Editor. During Surround track bouncing the abbreviations are automatically added to the resulting WAV files as the file name.

Example: Select the name "SurroundMix" when Surround bouncing a 5.1 Surround mix as 6 mono files. This will generate the following WAV files: „SurroundMix_L.wav“, „SurroundMix_R.wav“, „SurroundMix_C.wav“, „SurroundMix_Ls.wav“, „SurroundMix_Rs.wav“ und „SurroundMix_LFE.wav“.

Median / Frontal These columns describe the position of the respective loudspeaker in relation to the median and frontal levels.

Position X / Position Y: This describes the precise position of the speaker in a system of coordinates. You can edit the coordinates manually. The settings refer exclusively to the loudspeaker arrangement in Sound Field Mode.

Audio device: Assignment of a physical output on an existing sound card for the respective Surround bus. For each Surround preset you can save a default device setting (view page 299).

Lock channel position

If this is selected, the position of the loudspeakers cannot be changed using the mouse in the Surround panning module. This function is always initially activated and should only be switched off in special cases (e.g. variable speaker positioning).

Up arrow/down arrow buttons

If a line has been selected in the configuration table (corresponds to a Surround channel), this line may be moved up or down within the table.

The order of the Surround channels in this table determines the order for display of the mixer, Surround Editor (e.g. peak meter), and the Surround effects windows.

LFE Filter

You can activate a filter for the LFE channel in “Surround Setup”. You can choose between a 4-band EQ (default setting) and an FFT filter. These filters can be adjusted by clicking on “Edit...”.

Cut-off frequency: Here you can set the cut-off frequency for the low-pass filter.

High Frequency Dampening: The dampening of signal above the cut-off frequency is set to 12 dB/octave for the 6-band EQ.

Edit EQ: A dialog for the filter settings opens here.

Set All Tracks to Stereo Position

The downmix is factored in during a stereo master bounce project, i. e. you can stereo bounce from a surround project when the downmix (view page 322) is set.

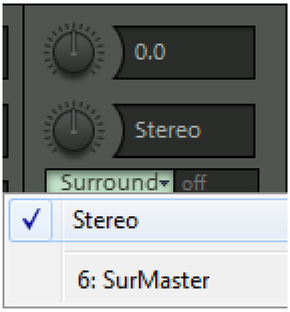
Surround panning

Now that the basic settings have been made, proceed to the actual mixing. There are two methods for mixing audio material displayed in the VIP window into the Surround master:

Track-based Surround Panning

In the Surround Editor (view page 303) of the corresponding channel strip in the mixer, each audio track in the arranger can be assigned to a position in the Surround panorama. All of the objects on the respective track will then be positioned at this location in the Surround panorama.

The Object Editor of this object displays the following the panorama setup:

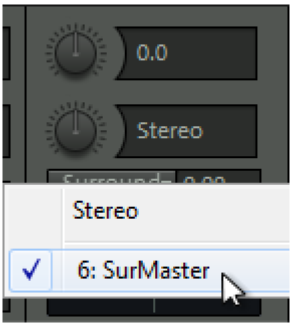


The selection "Stereo" under "Surround" allows you to route the object to the corresponding track, i.e. to use track-based Surround panning. In this case the audio signal runs through the channel of the respective mixer track. This is the default setting for Surround projects in the Object Editor.

Right-clicking in the Surround Panorama Field opens the corresponding Surround Editor (view page 303). Here you can set the positioning in the Surround panorama.

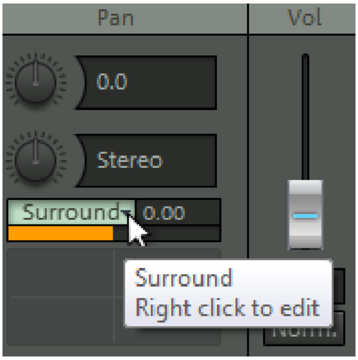
Object-based Surround Panning

In the Object Editor, each object can be routed to the Surround Master or another Surround submix bus. To do this, open the Object Editor of the object to be edited with a double-click and set the "Surround" button in the "Pan" section to "SurMaster".



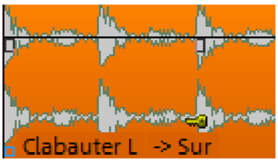
The setting "SurMaster" provides access to direct routing of the object to the Surround bus, i.e. to object-based Surround Panning.

An additional controller appears in the object editor which can be used to adjust the object's send level to the Surround Master.



Right-clicking in the Surround Panorama Field opens the corresponding Surround Editor (view page 303). Here you can set the positioning in the Surround panorama.

Object-based Surround panning is displayed in the object view in the arranger with the appendix "Sur".



Note: Object-orientated Surround panning will not route the audio signal through the channel strip of the corresponding mixer track. All settings made here (AUX send, EQ, etc.) have no effect on this object.

Track-based and object-oriented Surround panning gives you the freedom to choose between one of the two processes. If you hold down the "Ctrl" key while making a selection under "Surround" in the Object Editor, you can even select "Stereo" and "SurMaster" simultaneously and combine both types of Surround panning.

Surround Editor

In the Surround Editor the panning of a track or object on the Surround bus can be set and edited.

The following options are available for opening the Surround Editor for track-based Surround panning:

- Right-click the Surround Pan Display of the corresponding channel strip / track editor or
- Right-click on the "pan" button / panorama controller in the corresponding track header or
- Go to "Track" > "More" > "Pan/Surround Editor"

The Surround format and the track currently being edited are shown in the title bar of the Surround Editor. Example: "Surround Editor: 5_1 ITU - Track 4"

To open the Surround Editor for object-based Surround panning, double-click on the respective object to open the Object Editor. Right-clicking on the Surround Pan Display opens the Surround Editor for the respective object.

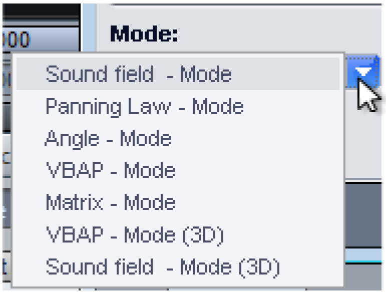
The Surround mode of the track currently being edited is shown in the title bar of the Surround Editor.

Display Elements in the Surround Editor

According to concepts of stereo tracks, a two-channel input meter is located on the upper right in Samplitude. Mono input signals are displayed in both scales of the input meter.

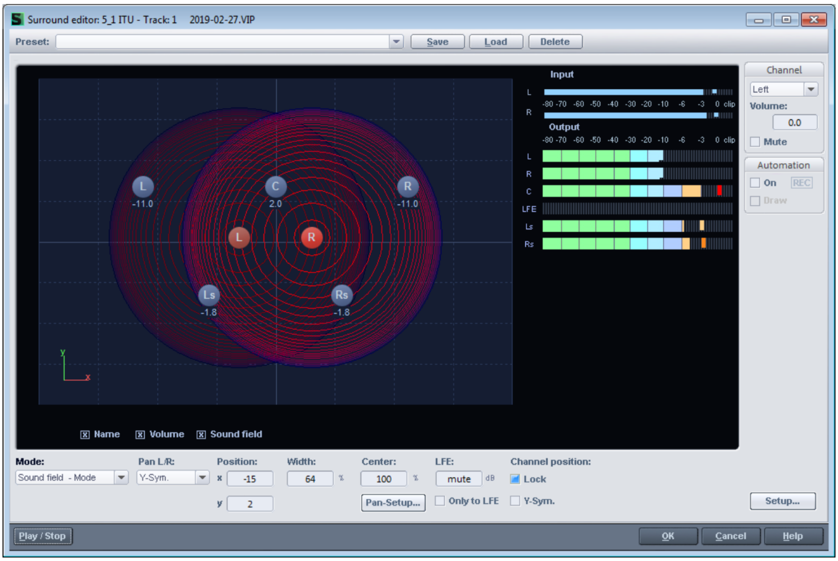
The level meters of the individual channels in the Surround Master are shown below it. Only the level components from the material currently being edited in the Surround Editor are displayed.

Under "Mode" you can choose from seven different display modes: Sound field mode, Panning law mode, Angle mode, VBAP mode, Matrix mode, and the 3D modes VBAP (3D) and Sound field (3D).



Note: The following remarks about the general display elements apply to all modes except the Matrix Mode. In addition, mode-specific parameters are explained in „Panorama Mode in the Surround Editor“ (view page 308).

The loudspeaker channels of the Surround setup are displayed as blue dots in the main panorama window. Their position in the Surround Panorama Field is dependent on the selected mode. If a channel is shut off using "Mute", it is displayed as a gray dot. The sound source to be positioned in the Surround panorama is displayed as a red dot. Depending on the "Pan L/R" setting, a stereo sound source can also be displayed with two dots.



The following information is also displayed under the main panorama window:

Name: The loudspeakers are labeled with the abbreviations specified in the "Surround Setup" window.

Volume: Level information is displayed for each loudspeaker. This displays the value of the signal component that is passed on to this bus by the sound source. The sum of the levels of both stereo sources is displayed for the stereo panorama setup under „Pan L/R“ (X-Sym., Y-Sym., XY-Sym., Parallel). If you hold down the "Shift" key and simultaneously click one of the two sound sources, only the level for this source will be displayed for a short time.

Sound field: Depending on the mode being used in the Surround Editor, the sound field is displayed either as a red surface or as concentric circles..

You can shut off the display information for "Name", "Volume" and "Sound Field" by deactivating the corresponding fields.

Pan L/R: In the Surround Editor, both mono and stereo sources can be positioned in the panorama. The function "Pan L/R" determines which way the mono and stereo signals will be arranged. For more information, please read the section entitled "Stereo and mono signal processing in Surround projects" (view page 313).

Position: In the Surround Editor, you can move the sound source with the mouse to the desired position in the panorama. Enter the position directly in the numeric fields "x" and "y" or change the numbers by moving the mouse pointer vertically on the numeric field or with the scroll wheel.

You can also move the sound source directly on the interface using the mouse. By holding down the following keys you can coordinate the position setting:

x + **Mouse movement:** Only a position change parallel to the x-axis is possible.

Result: L -> R movement

y + **Mouse movement or Z** + **Mouse movement:** Only a position change parallel to the y-axis is possible.

Result: Front -> Rear movement

c + **Mouse movement:** The distance from the sound source to position 0 (Px=0, y=0) remains the same. The result is a circuit.

Result: Circular movement

a + **Mouse movement:** The sound source can only be moved diagonally. A line running from the initial position of the sound source through point 0 of the system of coordinates determines the movements.

Result: Diagonal movement with consistent angles.

Pan Setup: (view page 312)

Center: This parameter controls the portion of the center channel for assignment of the sound source to the front channels. For certain applications (e. g. film sound) it is normal to reserve the center channel exclusively for dialog and to position music and background noises outside of it. A signal placed directly in the center is played back only by the center channel in 5.1 format if the center = 100%; at 0% this is only a phantom sound source that is output by the left and right channels. This parameter is often referred to as "divergence".

LFE(Low Frequency Effect): In this field you can determine the level component of the signal that will be routed to the LFE channel.

LFE only: The source is routed only to the LFE channel.

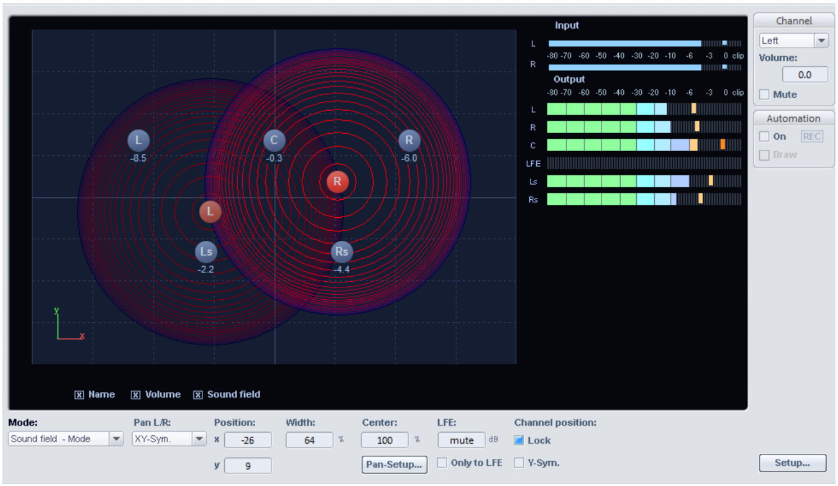
Channel: Mutes or changes the level of the bus outputs of the Surround Editor to the Surround buses. Deactivated bus outputs/speakers are displayed as gray dots in the panorama field.

Automation (view page 315)

Panorama Modes in the Surround Editor

Sound Field Mode

In Sound Field Mode the input signal is displayed as a concentric sound field. Each red line corresponds with a 3dB drop in the sound field level. The loudspeakers are positioned in such a manner that the distance from a single loudspeaker to a neighboring speaker remains constant. This positioning permits a uniform distribution of the sound source across all channels. Level ratios develop between the channels, which could not be achieved in other modes.



Example of use: Exact position localization, particularly during movements.

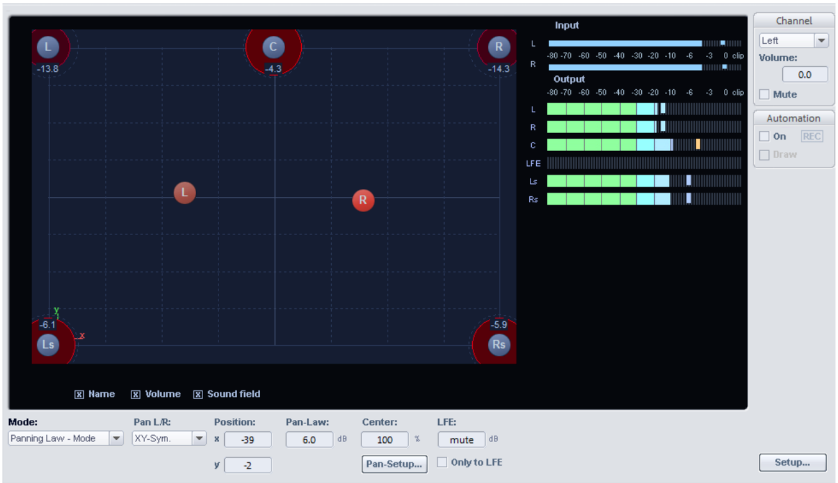
Level conflicts can occur for movement in a constant direction (e. g. a jet flying overhead). In this case, Angle Mode is better.

Width: This parameter influences the sound field width of the loudspeakers.

Pan Setup (view page 312) – Sound Field Characteristic: The sound field characteristic can be adjusted smoothly. "Invers. logar." produces a quick drop of the sound field and short fades between loudspeakers. "Logarithmic" provides a slow reduction of the sound field and longer transitions between speakers.

Panning Law Mode

This mode works with one of many digital mixing consoles in the usual Surround panorama display.



The position of the loudspeakers are displayed on the outer borders of the useable panorama. By clicking the "Sound field" option, you can visually display the graphical levels displayed across the Surround buses. The level distribution between two neighboring loudspeakers follows the -3 dB law, which means that a sound source directly in the middle of two neighboring loudspeakers will be emitted at a level of -3 dB.

Example of use: Static 2-dimensional panning, basic rough localization.

Because Panning Law Mode does not enable 100% exact localization, it is not suitable for dynamic panning (e. g. Automation).

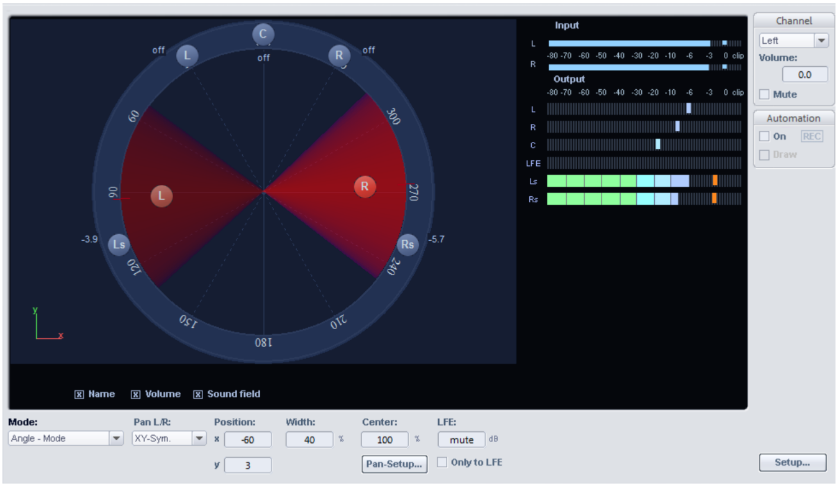
Pan Law: To compensate for volume fluctuations during panning you can also lower the volume of the pan mid position here.

Angle Mode

Here a sound field is displayed that radiates from the middle point of the circle. The sound source is located on the middle axis of this sound field. The loudspeakers are arranged on a sphere. The level components of a sound source on the respective channels are determined by the angle ratio between the sound source and the loudspeaker as well as the opening angle of the sound field. If the angle of the sound source and the channel perfectly match (i. e. the middle axis of the sound field is pointing directly at the loudspeaker: angle difference = 0), the level on this channel is the highest. The level on a channel drops as the angle difference increases.

Example of use: Good localization of direction of movements (e. g. jet flying overhead).

Angle Mode is not suitable for distance panning.



Width: This parameter determines the size of the sound field opening angle.

Pan Setup (view page 312) – Sound Field Characteristic: The sound field characteristic can be adjusted smoothly. "Invers. logar." produces a quick drop of the sound field and short fades between loudspeakers. "Logarithmic" provides a slow reduction of the sound field and longer transitions between speakers.

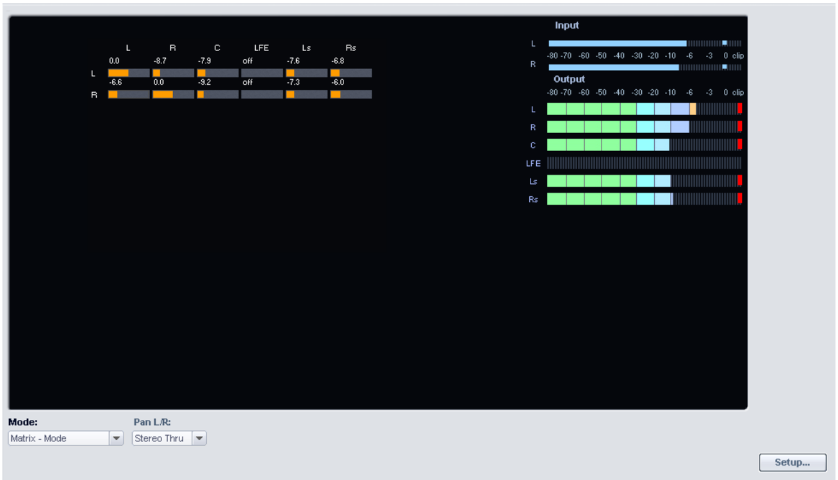
Pan Setup (view page 312) – Maximum sum output level constant: If this function is selected, the maximum level of all channel outputs of the Surround Editor will not exceed the set value. In Angle Mode, level drops in the sound field in connection with large opening angles can be avoided during movement.

Matrix Mode

In Matrix Mode you can directly enter the level that will go from the input signal to the individual Surround buses.

To enter the value manually, double-click on the numeric field. You can also set the level by dragging with the mouse over the level bar while holding down the mouse button. For very fine adjustments, hold down "Shift" key while doing this.

Examples of use: Analytical tasks like routing after track bouncing or simultaneous distribution of a signal over several Surround channels. Distribution over 3-dimensional alignments, e. g. 2+2+2 setup.



In contrast to the other modes, the settings from "Pan L/R" in Matrix Mode have the following meaning.

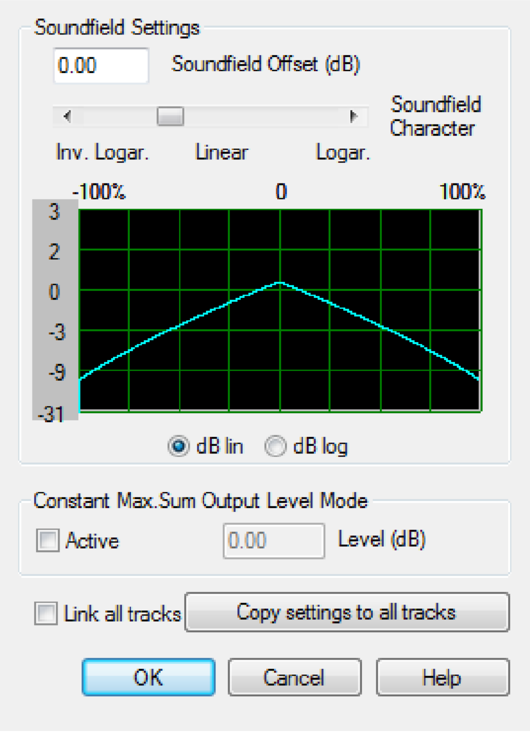
Pan L/R mono: The levels of the left and right input signals are applied equally.

Pan L/R stereo: Controls the levels of the left and right input signals individually.

The remaining "Pan L/R" settings do not apply. No reflective sound sources will result.

Pan Setup

In the "Pan Setup" dialog window you can make more settings for the Surround Editor. The dialog window can be opened by clicking on "Pan Setup" in the Surround Editor or right-clicking on the sound source.



Sound field offset (dB): The set offset is included for sound sources that are directly positioned and routed to a single Surround channel/speaker. For example, signals traveling from directly routed positions to panning between individual channels can be compensated if they are too strong.

Sound field characteristic (only available in the Surround Editor in Sound Field or Angle Mode): The sound field characteristic can be adjusted smoothly. "Invers. logar." produces a quick drop of the sound field and short fades between loudspeakers. "Logarithmic" provides a slow reduction of the sound field and longer transitions between speakers.

Maximum sum output level constant: If this function is activated, the maximum level of all channel outputs of the Surround Editor will not exceed the set value. This means that unintentional fluctuations of the max level can be counteracted as positions change. This function is particularly useful when you want to move certain sources through the move using automation.

Link all tracks: Changes in the following settings within the Surround Editor are always transferred to all tracks in the arranger: Sound Field Offset, Level Drop Characteristics, Output Volume Sum Constant, Center, Pan-law/Width, LFE level and settings, and channel settings such as level changes or mute.

Copy settings to all tracks: The settings made in this Sound Panorama Module (sound field offset, level drop characteristics, output volume sum constant, center, pan-law/width, LFE level and settings, channel settings such as level changes or mute) are copied to all other tracks once.

Surround presets

Save your own frequently used settings of the Surround Editor as presets here. In addition to the position of the sound source, the mode and the settings of Pan L/R are also saved.

With "Preset > Load" you can recall your presets. All surround presets are available also with a left click on the Surround pan display in the mixer. There are also a couple of pre-defined settings for frequently used pannings.

Pan L/R: Stereo and Mono Signal Processing in Surround Projects

Mono: For stereo signals a mono sum is created from the left and right channels. This is positioned as a single mono sound source in the panorama.

Mono signals are positioned as a single mono sound source in the panorama.

X-Sym: The following applies to stereo signals: The left and right channels are symmetrically aligned to the x-axis. For example, this enables you to pan a stereo signal to front L (Left) / LS (Left Surround).

The following applies to mono signals: A second (mono) mirrored sound source of this signal is positioned in addition to the original mono source.

The x-axis is the mirror axis.

Y-Sym: The following applies to stereo signals: The left and right channels are symmetrically aligned to the y-axis. For example, this allows you to pan a stereo signal to front L (Left) /R (Right).

The following applies to mono signals: A second (mono) mirrored sound source of this signal is positioned in addition to the original mono source. The y-axis is the mirror axis.

XY-Sym: The following applies to stereo signals: The left and right channels are symmetrically aligned to the x and y axes. For example, this enables you to pan a stereo signal to front L (Left) / RS (Right Surround).

The following applies to mono signals: A second (mono) mirrored sound source of this signal is positioned in addition to the original mono source. This mirrors the original source in relation to the X and Y axes.

Parallel: The following applies to stereo signals: The left and right channels are kept in constant distance from each other and are moved in parallel. The distance between the two sound sources can be changed when the Ctrl key is pressed.

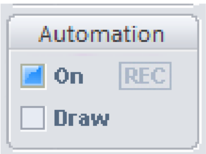
The following applies to mono signals: The original and mirrored sound sources are kept in constant distance from each other and are moved in parallel. The distance between the two sound sources can be changed when the Ctrl key is pressed.

Stereo Thru: For stereo signals, depending on the position of the sound source, the same level ratios are reproduced on different channels. However, only the left signal is used for all left channels, and only the right signal for all right channels, and the mono share for the center and LFE channels.

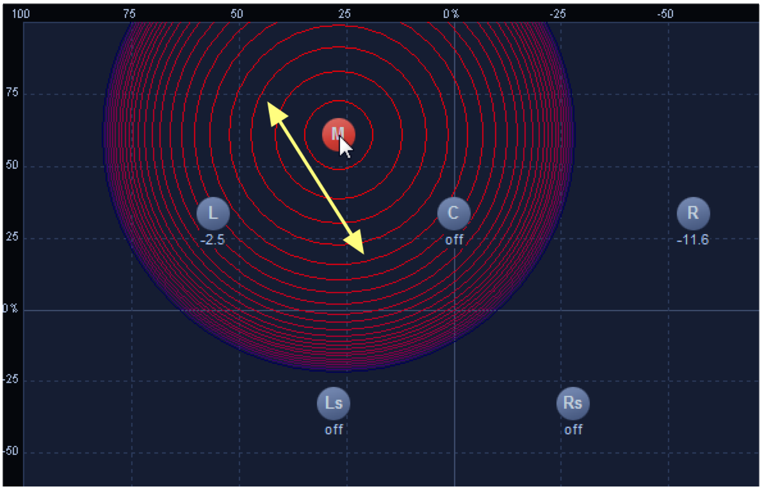
The following applies to mono signals: No special function, identical with mono mode.

Track-based Automation in the Surround Editor

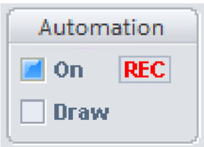
To automate panning movements, first activate the Automation option in the Surround Editor.



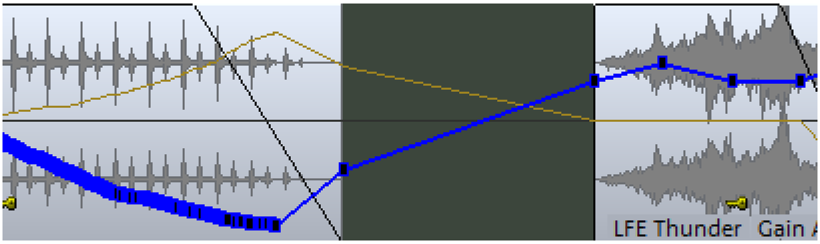
Now start playback. When you move the sound source with the mouse pointer during playback,



these movements will now be drawn



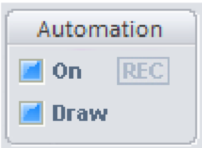
and displayed in the arranger as a curve.



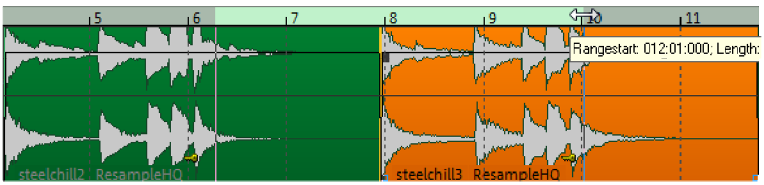
If the curve is not visible on the track, click on "Show" in the "Automation" section of the track editor to display it. You can edit the automation at any time by using the Automation Draw Mode in the arranger.

Drawing Surround Automation Curves

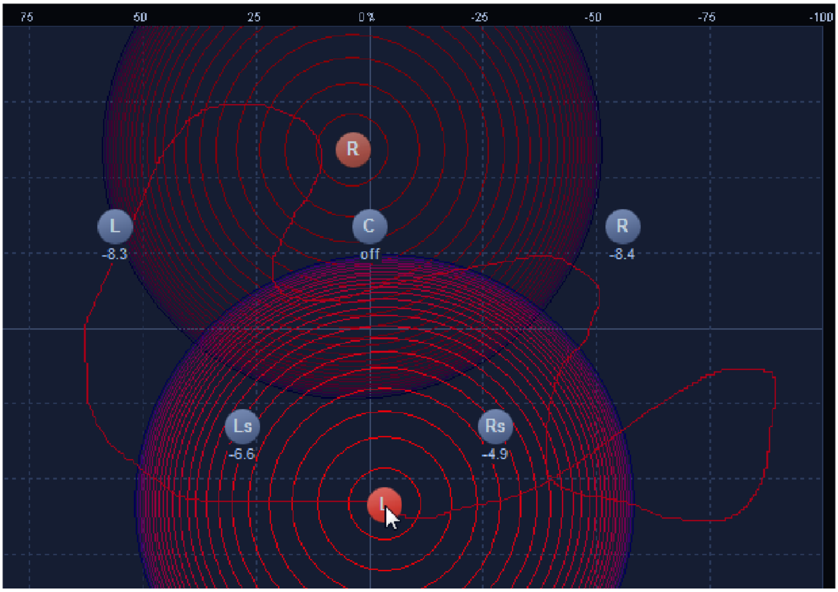
Activate the "Draw" option as well if you want to draw the automation directly in the Surround Editor.



First, in the arranger, select the track range where the automation will take place.



Now you can draw the movement progression of the sound source in the Surround Editor.



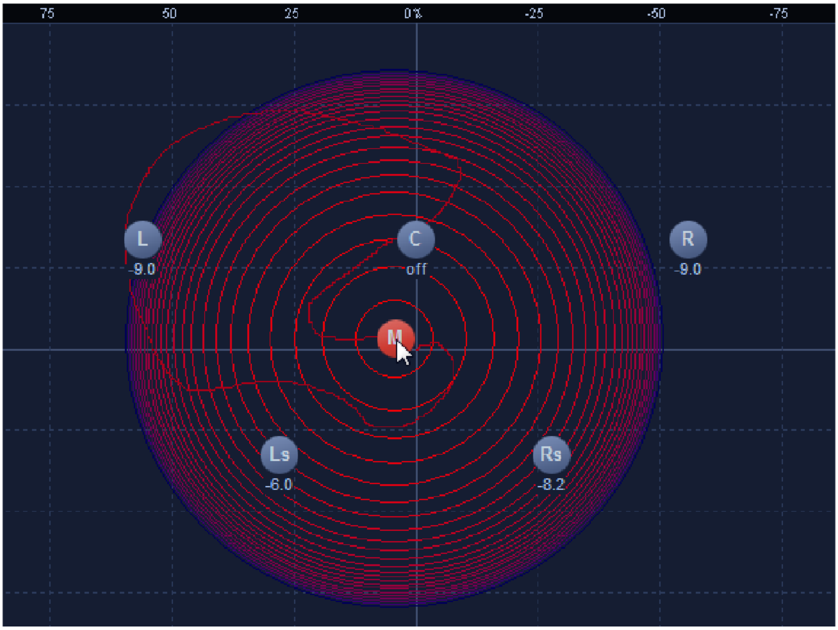
This progression will then be applied the next time the selected range is played back. Retroactive editing of the curve is possible here too with the help of the Automation Draw Mode in the arranger.

Object-based Automation in the Surround Editor

First open the Object Editor of the object to be automated. The setting "SurMaster" provides access to direct routing of the object to the Surround bus, i.e. to object-based Surround Panning.

Right-clicking in the Surround Panorama Field opens the corresponding Surround Editor.

Activate the "Draw" option as well if you want to draw the automation directly in the Surround EditorSurround Editorion of the sound source in the Surround Editor.



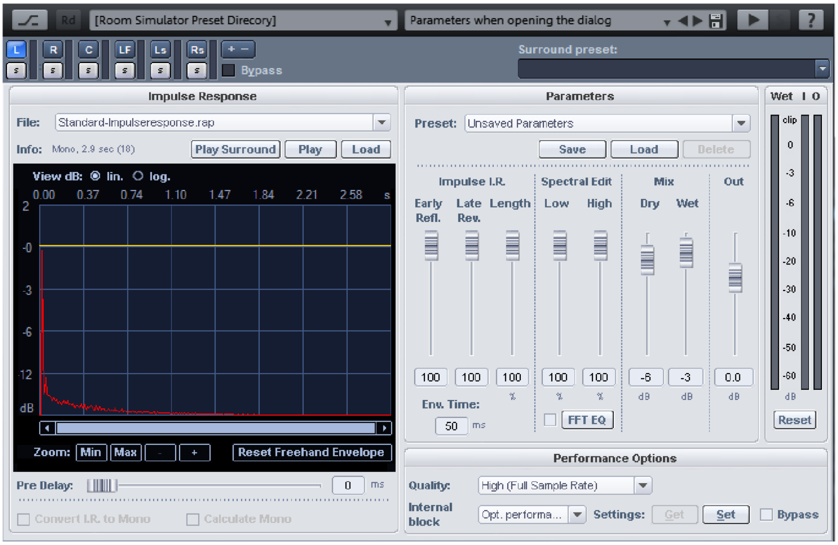
This progression will then be applied over the length of the object the next time it is played back. Retroactive editing of the curve is possible here too with the help of the Automation Draw Mode in the Object Editor.

Effects in Surround projects

Surround Effect Plug-ins

In Samplitude there are Surround effects available for all Surround buses which can be accessed through the "Plug-ins" section of the mixer. Here you can also add Surround-compatible VST effects from third parties. The surround channels can be grouped together to Surround Effect Groups.

- Multiband Dynamics
- Advanced Dynamics
- sMax11
- EQ116
- FFT Filter
- Room Simulator
- DeHisser SE
- Vocoder
- Delay



In Surround processing, each effect can be applied on up to 6 channels.

Surround Control Group

Surround control group The Surround channel buttons for each Surround mode are located in the Surround control group that appears with the Surround-capable effects above the effects dialog.

You can also switch the individual Surround channels to solo.

Grouping Mechanism for Surround Effects

Grouping and ungrouping:

To turn on Group Mode, click on the "+" button in the Surround Effect dialog. Now click on the buttons of the Surround buses that you want to group or ungroup. The corresponding Surround channels will now be marked with points of the same color.

If you click on "+" again to deactivate Group Mode, you can still open the created groups by clicking on the corresponding channels.

Adding Effect to Surround Buses: If you have grouped the Surround buses,

all the channels in the group that are open in the mixer will be calculated with an instance of the effect.

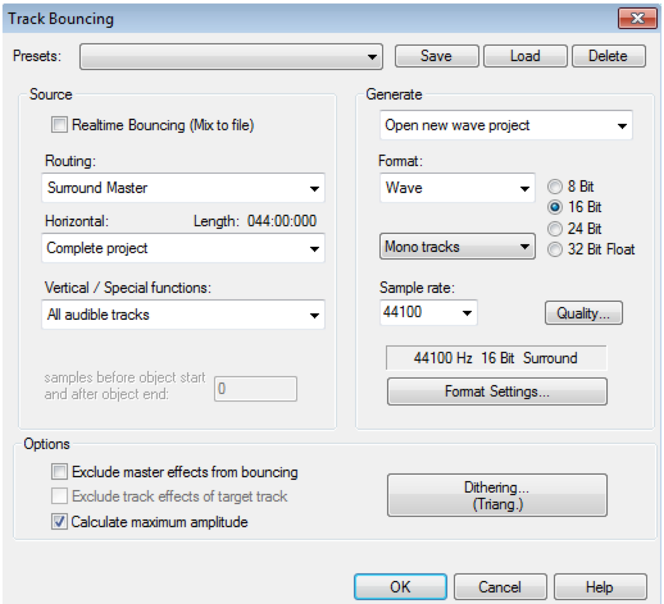
Effects of Grouping

- The parameters entered in the effect dialog are applied to all the channels in a group.
- For dynamic effects (Advanced and Multiband Dynamics), the control signals are compiled from all channels in the group, just as they are compiled from two stereo channels in the case of stereo.

Surround Bouncing

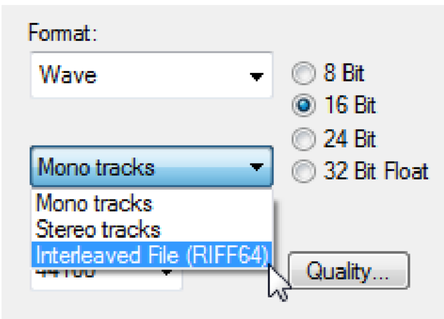
Surround bouncing enables you to compile a complex Surround mix. The outputs of the individual surround busses are used as a recording signal.

The "Track Bouncing" option can be accessed through the "File" menu.



Format: The following output formats are available for Surround bouncing:

- Wave files (Mono)
- Wave files (Stereo)
- Wave files (Interleaved RF64)



RIFF64 (also known as RF64) is a format for combining individual channels into a Surround sound production (e.g. for 5.1) as an interleaved file.

If you use the Stereo and Surround Master simultaneously in the mixer, you can select the setting "Surround + Stereo Master" in the "Routing" section of the track bouncing dialog to export RF64 files in the 5.1+2 format, for example.

The remaining settings can be selected during regular track bouncing (view page 582).

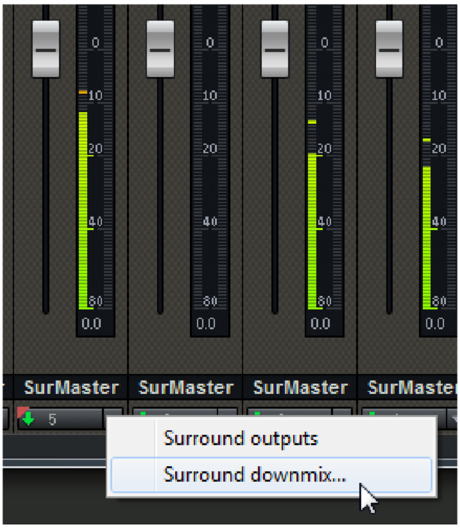
The abbreviations for individual Surround buses set in the "Surround Setup" (view page 298) will be added at the end of the selected names of files to be bounced.

Example: Select the name "5_1_Surround" when Surround bouncing a 5.1 Surround mix as 6 mono files. This will generate the following wave files: „5_1_Surround_L.wav“, „5_1_Surround_R.wav“, „5_1_Surround_Ls.wav“, „5_1_Surround_Rs.wav“, „5_1_Surround_C.wav“ „5_1_Surround_LFE.wav“. For stereo file output, the following files will be generated: „5_1_Surround_LR.wav“, „5_1_Surround_LsRs.wav“, „5_1_Surround_CLFE.wav“.

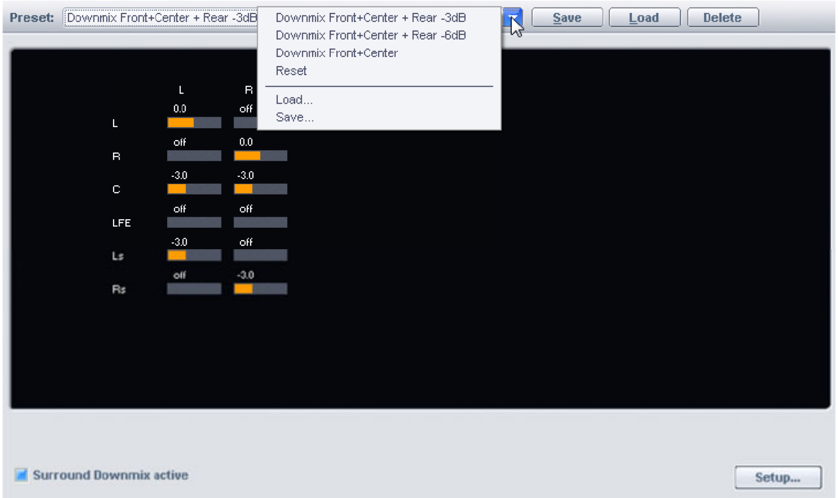
Surround up or downmix

A master bus can also be routed to any other surround or stereo master, while the required upmix or downmix is performed. For example, you can create a stereo mix on the stereo master from a project with a surround master or, in addition to a 7.1 surround mix, a 5.1 surround mix.

Open the output menu in the mixer of the corresponding Surround master and select the additional master while holding down the Ctrl key.



To open the downmix dialog where you can set the downmix coefficients, select the Master in the Output menu again.



In the preset list above there are common presets for downmixes of 5.1 in stereo. Additionally, you can also set and save your own downmix coefficients. The downmix is accounted for during Stereo Master bouncing.

MIDI in Samplitude

In this chapter you'll learn how MIDI data can be imported, played back and edited with internal editors in Samplitude.

MIDI data is also used to control internal and external synthesizers and VST instruments (view page 411) as well as the MIDI timecode synchronization (view page 450). Remote control using external hardware controllers also takes place through MIDI control signals.

> **Tip**: For seamless work with MIDI, always use the monitoring settings "Track FX Monitoring", "Hardware Monitoring/Hybrid Engine", or "Mixer FX Monitoring/Hybrid Engine".

MIDI setup

All global MIDI settings are provided in the system dialog (shortcut: "Y") under "System options" > "MIDI".

Please read "MIDI settings" (view page 81) for more.

Import, record, edit

Record MIDI Tracks

> To find out more about MIDI recording and MIDI recording modes, see "MIDI recording (view page 51)".

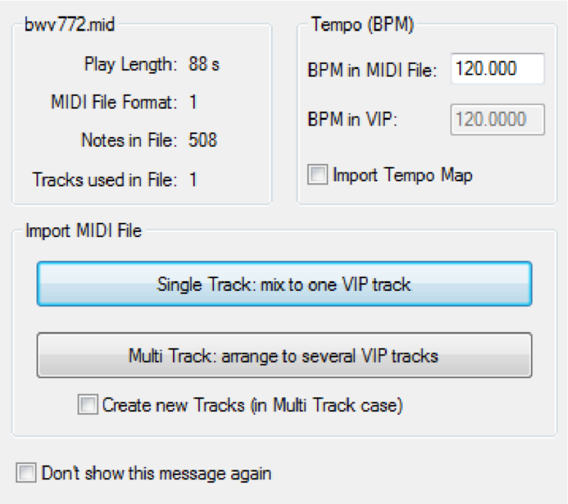
Importing MIDI Files

Existing MIDI files can be imported into a Samplitude VIP project as objects.

Menu:	File > Import > Load MIDI File...
Keyboard shortcut:	Shift + M
Mouse:	Drag & drop from file browser or Windows Explorer

When importing MIDI files, note that only type 0 and 1 standard MIDI files (SMF) are compatible with the import function. This corresponds with tracks featuring format 0 and multiple tracks featuring format 1. Each file should feature the file extension .MID in order to be detected as a valid import format.

The following dialog is displayed automatically:



Standard MIDI files often also contain tempo information. Samplitude displays this information in BPM (beats per minute) in a separate section of the dialog. At this stage of importing, select whether to adjust the virtual project tempo to the tempo of the imported MIDI file or not. Place a checkmark by the option "Transfer Tempo Map".

If you select the option "All MIDI channels into one VIP track", Samplitude inserts the MIDI object into the currently selected track of the virtual project. The MIDI object that is created contains all the tracks of the MIDI file.

To provide a separate track in the virtual project for each MIDI channel of a standard MIDI file (format 1), select the option "Each MIDI channel into a new VIP track".

Samplitude also offers the option "Create new tracks" for multitrack import of MIDI files. If this function is not selected, existing tracks will be used for importing.

Editing MIDI Objects

Editing MIDI objects in Samplitude follows the same principle as editing audio objects: MIDI objects can be copied, split and trimmed, have fade handles for fading in or out, and a volume handle that scales the MIDI velocity.

When MIDI objects are frozen („Object“ > "Freeze Objects (view page 709)“) the audio return signal of a software instruments replaces the MIDI object with audio objects.

Note: Make sure that the audio return signal of your MIDI receiver (VST or ReWire instrument) is routed to the MIDI track.

MIDI objects can also be edited using the MIDI object editor as well as in the different MIDI editors such as the Matrix editor, the drum editor, the controller editor, the event list and the note editor. The MIDI Object Editor will then only change the object properties and you can change the MIDI files yourself using the MIDI Editor.

MIDI object editor

Menu: Object/Object editor

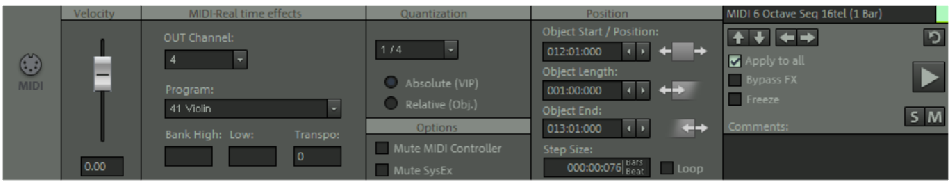
Keyboard shortcuts: Ctrl + O

Mouse: Shift + double-click

MIDI object parameters may be easily edited using the MIDI object editor which is structured similarly to the audio object editor. Among other options, you can set sound, length, volume, and program swapping for each MIDI object.

A Samplitude MIDI object can contain events in up to 16 channels. For example, it is possible to control a multi-output VST instrument using a single object on several MIDI channels.

Or select a specific MIDI channel for all events in the MIDI object editor.



You can also use the MIDI object editor to quantize MIDI files into MIDI objects.

Note: These settings influence the MIDI object in realtime and not destructively, making the changes invisible in the MIDI matrix or drum editor.

Important functions and settings in the MIDI object editor:

Velocity: Normally, every MIDI note contains a velocity value which determines how "hard" the note will be played back. Changing the velocity value in the object editor scales the velocity of the notes in the MIDI object to its physical maximum or minimum values (1 or 127). Changing the fader is identical to changing the top-mid handle of the MIDI object in the VIP track.

MIDI real-time effects: The settings for a MIDI channel can be found here:

OUT channel: Here you can redirect the MIDI data of a channel to another channel.

Program: The values preset here will always be sent if the object is played back again.

Bank High/Low: This sets the MIDI bank change bytes.

Transp.: This button transports all MIDI notes.

Quantization: While the quantization conducted in the MIDI editor has an immediate effect on the start position of the notes, a virtual quick quantization takes place. Absolute refers to the absolute resolution of the VIP project while Relative refers to the relative resolution of the object itself.

Options:

Mute MIDI Controller: Switches the MIDI controller off.

Mute SysEx: Prevents the sending of SysEx data.

Position: Define the object start/position, object length, object end and step size here. The setting is equivalent to moving the object in the VIP track or changing the length using the lower length handles right and left in the object.

Loop: If you set a check here, then you will set the MIDI object to "Loop" mode. Now drag the bottom right object handle to the right and you will see that the events of the MIDI object are arranged in a row, according to the new length of the MIDI object.

The object name is displayed in the upper text field and can also be edited.

In the square field to the side, you can select the object color.

Use the up/down arrow keys to "jump" up or down to the object of a neighboring track.

Use the double arrow buttons to jump to the previous or next object on the same track. If multiple objects are selected, these buttons will not be active.

If the "To all" option is active, all selected objects are updated when the object editor is opened. All recently applied settings that have been made in a selected object will be applied to all other selected objects.

Note: In a MIDI object - velocity changes are transferred relatively, i. e. they are increased or decreased relative to the velocity of other selected objects.

Bypass FX: Here you can deactivate all effects.

Freeze: If you click on the freeze function for an object, this will be calculated as a new wave file with all the associated effects. The new freeze object takes the place of the original object.

Play/Stop: This button corresponds to the normal playback function (spacebar).

Lock: The "Lock" button has the same function as the key button in an object. The object is then protected against any inadvertent horizontal movement. Additional locks against vertical moving, volume adjustment, fades, length adjustments, ripples or deletion can be set in the system options (shortcut: "Y") under "Program" > "Object lock definitions" (view page 692). Locking can be temporarily cancelled by pressing the "Alt" key.

Solo: During playback this button moves the playback marker to the start of the selected object and plays that object only.

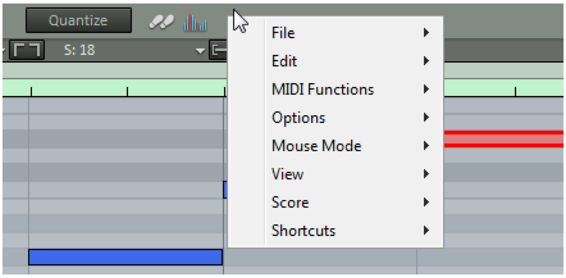
Mute: This button mutes the selected object. By right-clicking you can also mute the left and right channels individually.

Comments: You can type comments relevant to the selected object in this field.

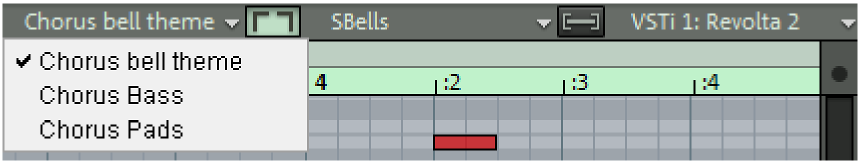
Opening the MIDI Editor

The MIDI Editor is opened by double-clicking on a MIDI object in the project. You can also start the MIDI Editor for all selected MIDI objects by selecting "Object > MIDI Editor ..." in the menu or by clicking on the "MIDI Editor" button.

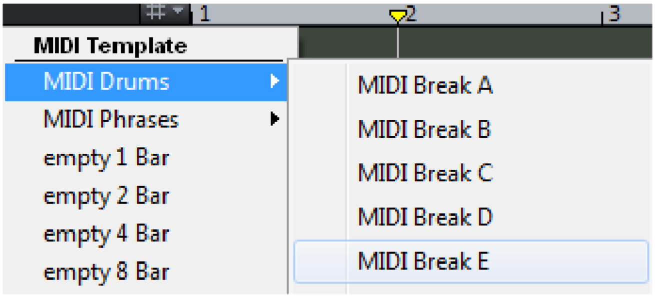
If you open the MIDI editor in the Docker (view page 90), the MIDI menu can be opened by right-clicking on an open area in the toolbar.



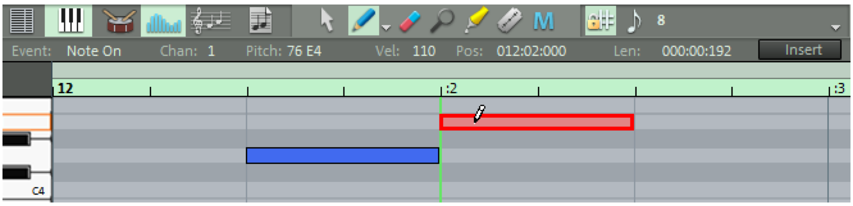
The MIDI Editor always displays all of the selected MIDI objects. In the field to the right you'll see the MIDI output and the software instrument that is activated for this track.



If no MIDI object is selected, a window appears which lets you create an object at the playback position. You can then choose between a menu of predefined MIDI templates from within the Samplitude “Templates” sub-folder.



Now you can use the Draw Tool to create events in the MIDI editor (view page 332) or record notes using your MIDI keyboard.



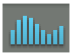
Detailed information about MIDI recording is provided in the chapter "Samplitude Quick Start -> MIDI recording (view page 51)".

Working with the MIDI Editor

MIDI data can be edited in the MIDI Editor in five sub areas:

 Matrix Editor (Piano Roll)

 Drum Editor

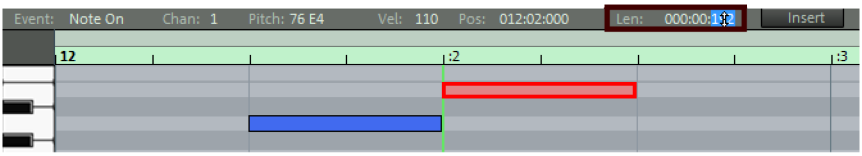
 Controller Editor (for example, velocity, MIDI volume...)

 Event List

 Score Editor

Various tools are available for editing, e. g. the pencil or eraser tools.

The "current" MIDI events displayed in bright red can be very precisely adjusted in the editing fields above the editing window (view page 335).



Changes such as moving or deleting notes always apply to all selected MIDI events (displayed in red). Changes to the selection in an editor range always apply to every other editor range as well. For example, you can select a group of notes in the Matrix Editor and then change the velocity of these note groups in the Controller Editor which modifies all selected notes simultaneously.

MIDI Editor: Select events, delete events

(Matrix Editor, Drum Editor, Controller Editor, Event List, Score Editor)

- **Select events**: Left click the event or drag out a frame around the event while pressing the left mouse button
- **Add/remove event to/from selection**: "Ctrl" + left click on event or "Ctrl" + pull the frame over the new event
- **Set or change current events within a multiple selection**: Left-click on selected event
- **Set current event, deselect all other events**: Double-click on event

- **Selection of events within a range**: Left-click on the first event, then left click on the last event while holding down the Shift key. Another alternative is to drag out a frame around the events you want to select.
- **Selection of all notes of a pitch**: Double-click on the assigned key on the keyboard display.
- **Select all notes behind click position**: Shift + Alt + Double-click
- **Select all notes**: Ctrl + A
- **Freehand drawing and changing of note lengths without snap points**: Hold Alt key down while dragging.
- **Select previous or next note**: Left or right arrow keys
- **Delete events**: Right-click
- **Delete selected events**: Backspace or Del.

MIDI editor: Copy/duplicate events

You can copy selected events using "Ctrl + C" and then paste them at any playback marker position using "Ctrl + V".

Copying and pasting MIDI data is available not only within the MIDI editor, but also between different MIDI objects. Copied MIDI data is always inserted at the current position of the playback marker.

Duplicate: Press "Ctrl + D" to copy selected notes and add these to the next snap point behind the selection. Activate the quantization grid ("MIDI menu -> Quantization grid active") to do this.

MIDI-Editor: Move events

To move events, move the mouse over the event. The mouse pointer turns into move icon. Now you can move the event. If you have snap turned on, the events move horizontally according to the snap quantization values.

To temporarily remove a event from the snap quantization, hold the "Alt" key when moving.

If you only want to move an event horizontally, avoid accidental vertical movements by simultaneously holding the key "H". The move symbol will turn into a horizontal double arrow.

If you only want to move an event vertically, avoid accidental horizontal movements by simultaneously holding the "Shift" key. The move symbol will turn into a vertical double arrow.

MIDI Editor: Zoom use the scroll wheel

Just like in the project window, you can set the vertical and horizontal picture detail as well as the zoom factor in the MIDI editor using the scroll wheel:

Scroll wheel: Horizontal scrolling

Shift + Scroll wheel: Vertical zoom

Shift + Ctrl + Scroll wheel: Vertical scrolling

Ctrl + Scroll wheel: Horizontal zoom

You can change the scroll wheel settings at any time by going to "File > Program Preferences > "Edit Keyboard Shortcuts and Menu (view page 626)".

Synchronized image view in Arranger and MIDI editor:

If you change the horizontal display using the scroll bars, holding the Shift key changes the corresponding Arranger window.

MIDI Editor: Mouse Mode/Toolbar

There are many different editing tools available for event creation and editing within the MIDI editor. For every tool (except the eraser), editing events functions in the same way as described above. The tools behave differently only when clicking in free areas. Access the mouse modes/tools in the MIDI editor "Mouse mode" menu and with the following buttons:

Selection (keyboard shortcut: 1)

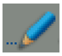
Lasso: Hold down the mouse button to draw a selection window.

Clicking on a free range highlights an existing selection.

Draw (keyboard shortcut: 2)

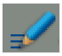
Draw an event by clicking the selected MIDI editor and dragging to the right with the mouse. The event snaps according to the current grid quantization values. If the "Alt" key is held down, the event can be drawn freely (without the snap function).

Drum Pencil (keyboard shortcut: 3)



A sequence of notes can be drawn with this tool. The length of the notes and the note intervals are determined by the current quantization settings. Moving the mouse back (to the left) while holding down the mouse button removes previously drawn notes.

Pattern Pencil (keyboard shortcut: 4)

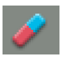


Select the MIDI events to use for the pattern first, then save this pattern using the keyboard shortcut "Ctrl + P". Now you can draw a sequence of previously selected patterns with the Pattern Pencil. Moving the mouse back (to the left) while holding down the mouse button removes previously drawn notes.

Adjust Velocity (keyboard shortcut: 5)

When this tool is active, drag the mouse vertically and the velocity values of the selected events will be increased or decreased relative to each other. Holding down "Shift" changes all velocity values absolutely, i. e. every changed event has the same velocity value after editing. To recognize the velocity of events by their levels, switch on "Velocity" for a controller slot button in the Controller Editor Range.

Eraser/Delete (keyboard shortcut: 6)



This deletes the MIDI notes by left-clicking. If multiple notes have been selected, they will all be deleted. The Eraser Tool can also be used on specific MIDI notes to "rub them out" and delete them. The eraser is also available for all other tools (except the magnifying glass) by right-clicking.

Magnifying Glass (keyboard shortcut: 7)

Zoom into the horizontal lengthwise display using the left mouse button; use the right mouse button to zoom back out.

The combination of left mouse button + dragging zooms in on the range. The zoom mode can be temporarily accessed again by holding the "Z" key at any time. After releasing the key, the set mouse mode is reactivated.

Glue Notes (keyboard shortcut: 8)

If you click on a MIDI note with this tool, the note will join with the next MIDI note of the same pitch.

Split Note (keyboard shortcut: 9)

This tool separates a note on the matrix into two notes. The separated note snaps to the next snap point.

Mute (keyboard shortcut: M)

In this mode you can mute or reactivate individual notes or selected groups of notes by clicking on them. This function is also available as a command in the MIDI editor menu under "MIDI functions" ("Ctrl + M").

Select Velocity Controller: The "Mouse Pointer" button in the MIDI editor toolbar activates the Combi-Tool (view page 361) which you can use to select notes, change values, draw lines and perform freehand drawing.

Draw Velocity/Controller: This activates freehand drawing (view page 361). In the controller editor the mouse pointer turns into a pencil. Detailed envelopes, ramps, and grades can be created by dragging in the respective controller range.

Draw Velocity/Controller as a line: This activates line drawing (view page 361) In the controller editor the mouse pointer turns into a cross. Create envelopes, ramps, and grades by dragging horizontally.

Tips:

- "Shift to pencil": The "Shift" key is now the hotkey for "Draw" mode. This does not apply to the magnifying glass mouse mode, since "Shift" activates the vertical zoom in this case.
- The keyboard shortcuts to switch the mouse mode/tool can be freely defined in the MIDI editor menu "Shortcuts -> Edit shortcuts".
- Delete Mode can be activated anytime by clicking/dragging with the right mouse button. For instance, use the pencil to insert new notes by left-clicking, or remove already inserted notes by right clicking (without having to change tools).

- Notes created in "Draw" modes contain the MIDI channel and velocity of the editing fields under the toolbar.
- Select the preceding/following notes with the left/right arrow keys. Change the pitch of the selected notes gradually with the up/down arrow keys.

MIDI editor: Editing fields

Editing selected events (editing fields): The properties of every event in the piano roll, drum editor, controller, and listen editor can also be edited with the editing fields under the editing tools. The following buttons are available for each note:

- Channel
- Pitch (byte 1)
- Velocity (byte 2)
- Starting time in bars:beats:ticks
- Length in bars:beats:ticks

The display of the ticks has a resolution of 384 PPQ. 384 ticks correspond to a quarter note.

Hold the left mouse button in the desired field and drag upwards or downwards to increase or decrease the value. The value steps are greater if you press "Ctrl".

This value may also be edited numerically with the keyboard. Double click in the editing field you wish to change and enter the value via the keyboard.

A few special properties result when multi-selecting events:

For the "Pitch" and "Level" parameters, you can change the values relatively by dragging with the mouse, or by entering the values numerically and then confirming with "Enter". You can make absolute changes for all selected events by simultaneously holding "Shift" while dragging with the mouse in the editing field, or by finishing the entry process with "Shift + Enter".

For the "Time" and "Length" parameters, you can change the values relatively by double clicking the desired value field and then confirming with the mouse wheel.

Note: MIDI channel changes are always permanent for multiple selections.

MIDI Functions

The commands in the "MIDI functions" menu of the MIDI editor always affect all note events in a particular range or all of the selected events. If no notes have been selected, the functions are applied to all notes.

Legato: Notes can be prolonged and played back in bundles.

Shortcut:	Ctrl + L

Quantize Notes (standard): This command conducts standard quantization of the lengths of the MIDI notes of all selected MIDI objects according to the MIDI quantize settings (view page 342).

Shortcut:	Ctrl + Q

Advanced Quantization:

Start Q: This command quantizes the start positions of the MIDI notes for all selected MIDI objects according to the MIDI quantize settings. The grid view corresponds to the start quantization value.

Start and Length Q/MIDI Quantization (Start and Length): This command applies quantization to the start and length (view page 343) of the MIDI notes of all selected MIDI objects according to the MIDI quantize settings.

Soft Q (quantize approximation): This command includes the current Soft Q level value in the quantization options (view page 346).

Q Length/Quantize MIDI Length: This command applies quantization to the lengths (view page 343) of the MIDI notes of all selected MIDI objects according to the MIDI quantize settings.

Quantize Note Ends to Grid: This command quantizes the ends of the MIDI notes of all selected MIDI objects according to the MIDI quantize settings.

Undo Quantization: Use this command to reverse all completed quantization steps.

Note: "Undo Quantization" is still possible after a VIP has been saved or loaded. During quantization, the distance to the next grid position saved together with the note and recalculated after "Reverse quantization" is selected. This way the original groove or phase of a note can be restored even after displacement or copying.

Quantization settings: In this dialog you can make detailed settings for the quantization (view page 342).

Quantize/Reduce controller: Controller values can be quantized or reduced with this function.

Humanize: This command factors in the current "Humanize" level value in the quantization options.

Mute Notes (Mute): Mutes and reactivates notes or selected note groups with a click.

Shortcut:	Ctrl + M

Remove Overlaps (polyphonic): Notes can be shortened to avoid any overlaps. Chords (simultaneously played notes) are recognized and left uncorrected, i. e. chords are not split up.

Delete Doubles (remove overlaps monophonic): If necessary, notes are shortened to prevent any overlaps.

Transform sustain pedal into note lengths: You can transform sustain values of events into notes lengths and display them in the MIDI editor.

Transposition: Here you can transpose the pitch of selected events by half steps.

MIDI Time Stretching: With this function you can make various changes to selected events, e.g.

- double the tempo
- halve the tempo
- scale to range length.
- stretch manually. This allows you to enter individual values in the Stretch Factor field.

Reverse: With this function you can reverse the playback sequence. This means that the sequence of notes will be played backwards. The result is a mirror image at the vertical axis.

Mirror Melody: This function mirrors the selected events in terms of the pitch of the current note. The effect is that each interval in the note sequence that was directed upwards will now be directed downwards and vice versa. The result is a horizontal mirror image.

If you don't define any particular notes, the mirroring will affect the mid pitch of all notes.

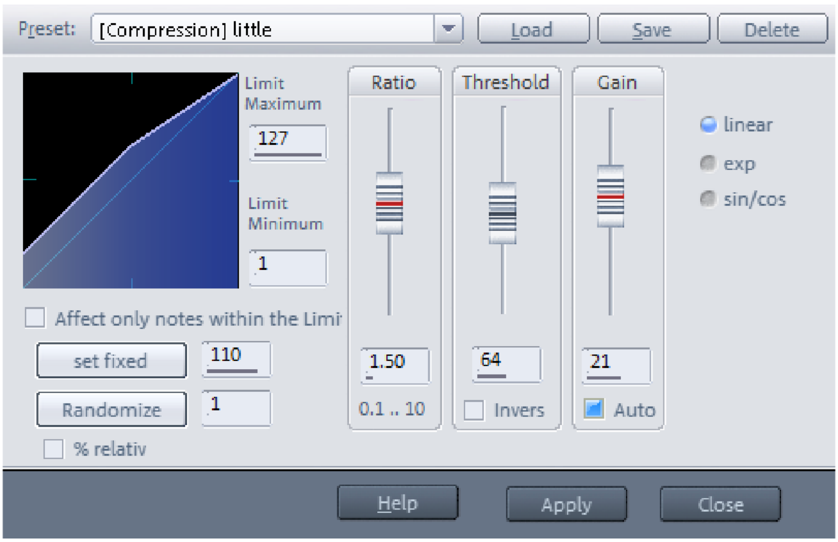
Velocity Dynamics

The MIDI velocity dynamics can be accessed as an offline effect via "MIDI editor MIDI Functions > Velocity dynamics", or as a real-time track effect in the track editor.

"Velocity dynamics" is a MIDI effect that edits the MIDI velocity dynamic of the selected MIDI notes. It can be used in real time as a track effect during playback and for MIDI-Thru (button in the MIDI section of the Track editor), and also offline via the corresponding menus.

Velocity dynamics makes it possible to adjust the MIDI velocity to the dynamics of the connected synthesizer, i.e. to adjust MIDI synths or VSTis, thereby compressing or expanding the velocity strength of the selected events.

Each input velocity value is assigned to a specific output velocity via the characteristic. The characteristic curve may be set linear like a compressor/expander, but other characteristics patterns are also available to produce exponential and sinus/cosine curves.



The Preset field contains a series of presets for different compressor or expander applications; these may be used to form the basis of individual adjustments.

Ratio (0.10 - 10.0): This parameter describes the level of compression/expansion as the MIDI input signal reaches the threshold.

Threshold (0 - 127): Threshold specifies the level at which MIDI events are adjusted via velocity dynamics.

Inverse: Velocity values above the thresholds aren't processed, but rather only those values below the threshold.

Gain (-128 – +128): The gain controller sets the amplification factor for events after they have been processed by other parameters

Auto: If this option is activated, the gain parameter will be adjusted automatically to reach a consistently full overdrive (Velocity 127).

Limit maximum and **Limit minimum** are used to limit the top and bottom velocity ranges by entering a minimal or maximum velocity value.

Offline mode

Offline mode makes the Set value button available. If this is pressed, then the selected MIDI events may be set to the velocity value entered in the field next to it.

Change velocity with the random function

With the Random Variation button you can assign several of the same velocity values to subsequently fluctuate around mean velocity values, e.g. in order to make a programmed sequence sound as if it were played live. In the field next to the button, enter the maximum deviation value allowed. The velocity values will be altered randomly within the range specified.

If **randomized variation should be displayed in relative %**, then this option must be checked accordingly. If the Randomize button is clicked now, the selected MIDI velocity values will change accordingly. At a lower dynamic in the original material, variation will be softer than positions with higher dynamics.

Only for notes in limit zone: A check placed here modifies notes with a velocity within the limit minimum and limit maximum. This enables all notes with a velocity of 100 to be set to 77 or only notes above velocity 100 to work with random variation.

Note: By setting the limit minimum and limit maximum to the same value, a fixed value may be set for the velocity for real-time use.

Quantizing

Irregularities during recording can be evened out with quantization. Samplitude features audio quantization as well as MIDI quantization.

Details about audio quantization are provided in the menu reference "Object" > "Quantize" > "Audio Quantization Wizard" (view page 694).

Recorded MIDI events can be set to the exact note start and note length values using MIDI quantization. Quantization can also be varied with additional parameters like "Soft Q", "Swing", or "Humanize", which adjust quantized MIDI notes to the musical properties of each song.

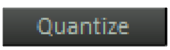
The quantization functions are featured in the MIDI Editor via the "MIDI functions".

The settings for MIDI start and length quantization can be found in the "Menu Object > Quantize > MIDI Quantize settings" or in the "Grid quantization value" and "Length quantization" fields of the MIDI editor's toolbar. If you enter the value # in the length quantization values input field, the value will be linked to the respective snap quantization value.

The setting "Beats" means that time signatures based on quarter notes serve as a quarter snap unit, and time signatures based on eighth notes serve as an eighth snap unit. The snap therefore follows the time signature and observes any time signature changes.

The grid view in the MIDI Editor corresponds to start quantization value. If you also hold down the "Alt" key, the grid will be disabled temporarily.

Clicking the "Quantize" button at the top right in the MIDI editor quantizes the start positions of all selected MIDI objects according to the MIDI quantize settings (view page 342). If no notes are selected, all notes will be quantized.



Shortcut:	Ctrl + Q

Right-clicking this button opens the dialog for the global grid/quantize options (view page 342).

The command "Extended MIDI quantize (start and length Q)" applies quantization to the start and length of the MIDI notes of all selected MIDI objects according to the MIDI quantization options.

"**Soft Q (approximate quantization)**" includes the current soft Q values in the quantization options.

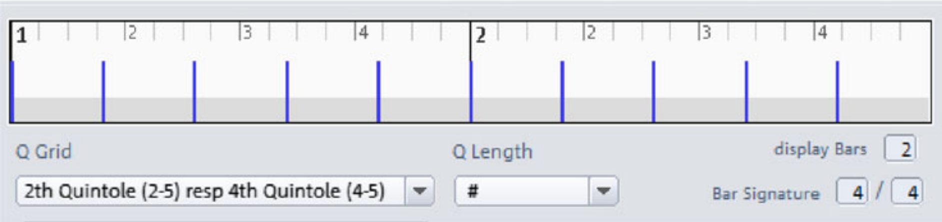
Quantize Note Ends to Grid This option can be used to extend the ends of the selected notes to the next snap point according to the set values.

Reset quantization: This function resets all quantization processes.

> **Note:** Press the Alt key to temporarily deactivate the quantization grid so you can adjust notes freely. The same applies for moving objects in the arranger window as well.

Quantization - window view

The quantization snap grid (Q snap) is displayed graphically, and the Q range appears gray. Only notes or slices within the Q range will be changed.

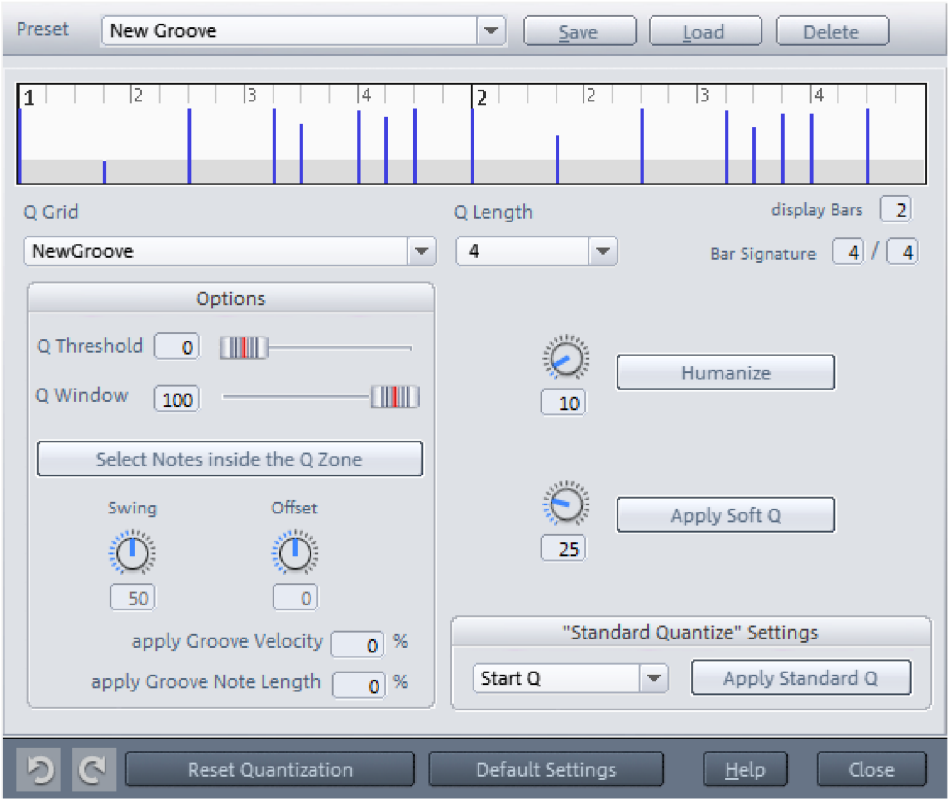


The beat names in the quantization grid depend on the beat signature that is set. The blue lines depict the effective Q snap as well the velocity level (bar height) for groove templates.

> **Note**: The beat signature in the quantization dialog is independent of the tempo map.

Quantization settings

For precise specification of the global grid and quantization options, a separate dialog is available to you in the menu under "Object > Quantize > MIDI Quantize settings":



This dialog can also be accessed in the MIDI editor via "MIDI functions -> extended Quantize > Quantize settings" or by right clicking on the "Quantize" button. This dialog displays the main view depending on the settings in the quantization area and the respective quantization snap points.

Settings applied here also affect the behavior of the "Input Q (view page 95)" button in the track editor.

In this case, MIDI quantize settings changed in the VIP are applied as a preset for new VIPs.

Quantization presets

Here you can select from a number of presets:

- **5-tuplet:** Quantization occurs in fifths

- **Magnetic quantize:** The "Q window" value is set to "50", i. e. only 50% of quantization will be considered. Only events that are within a range of 25% of the grid width left and right of the grid point will be quantized.
- **Swing:** The swing parameter is set to "75", i. e. in contrast to the binary rhythm, which features a "swing" value of "50", inclined/un-highlighted counting times will be set to delay. This results in a swing feeling
- **Triplets:** Quantization occurs in triplets
- **16th offbeat:** The quantization grid's timing is moved back a 16th note
- **8th offbeat:** The quantization grid's timing is moved back an 8th note
- "**New groove**" and "**More life for hi-hat**" provide groove templates

Of course you can create and save your own settings as presets.

Q grid/Q length

Both fields for "Q snap" and "Q length" match the fields in the MIDI editor's toolbar. The settings for grid and length quantization are also located here. The value "#" in the length parameter couples the length quantization value to the respective snap quantization value that has been set.

Tuplets (8th, 5th, 7th)

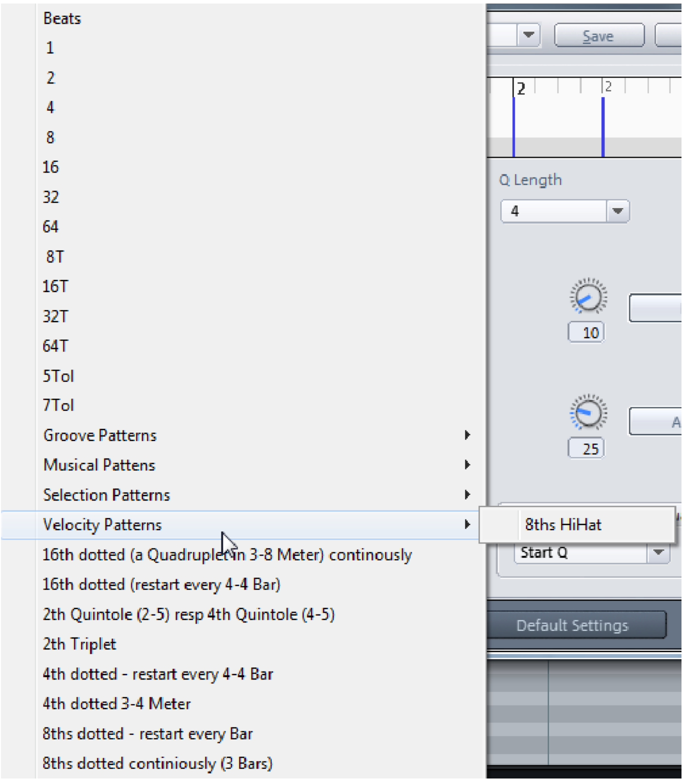
The Q grid also provides n-tuplet quantization values (for thirds, fifths, and sevenths). For example, if 7 is entered, the grid will be adjusted for septuplets. The count will be divided into 7 subsections for quantization. A thirds snap is indicated by a "T" behind the number.

Groove template

You can set a rhythmic pattern on all selected MIDI events or Audio objects with Groove Templates. Groove Templates adjust the selected range of a recording to a specific musical grid. So you can bring static MIDI patterns to life, quantize them according with available audio drum loops or create special metrics like dotted grid values. It is also possible to make a rhythmic pattern for the MIDI pen. As well as the position the velocity and note length can be also be changed with the Groove Template (only for MIDI).

A Groove Template enables a freely-definable grid. As a rule it is from one to four beats long and repeats itself in a loop. The length of the Groove Templates is in principle, however, freely-definable. You can also create a grid based on a track's bass drum, to draw a perfectly in-time bass using the MIDI pen. Length and start of the Groove Templates are continually set to all the beats.

You can select the grid from the Groove Template selection box, which puts binary, triplet and dotted note values at your disposal.



If a you have selected a groove template, the "Snap/Quantization value" field of the MIDI editor will indicate "Grv".

The MIDI editor may be used to create new groove templates yourself by selecting the MIDI events as the template and then accessing "Edit -> Create groove template from selection".

This quantizes the Groove Template to entire bar borders. So if you want to create a Groove Template that is 2 eighths long, choose a 2/8 time signature.

Additional to each note start position both the note length and velocity are saved in the Groove Template. In the file selection box you can give your Groove Template a name. Subsequently the Groove Template is immediately active and appears in the grid quantization list.

Groove velocity (appears only for groove templates): The groove velocity indicates in % how much the velocity information affects the groove template.

Groove note length (appears only for groove templates): The groove note length indicates in % how much the note length information affects the groove template.

Note: If a groove template is selected, then the parameters "Swing" and "Offset" will be deactivated and grayed out.

For audio objects, you can create groove templates by specifying the transients (AQ markers) for the audio object intended to serve as the template via "Object > Quantize > Extended audio quantize" and then accessing the "Create Groove Template from Transients" function. You can also reduce the range in which you want to create Groove Templates through another active range. Then only the AQ marker within the highlighted object range will be taken into account.

If you rename the new Groove Template and put it in the "fx-preset/Grooves" program folder, it will be available in the future in the Groove Template selection box.

Show Grid/Time Signature

Displays the desired beat signature and number of beats. The display window will change accordingly.

Q Threshold

The parameter "Q threshold" may be used to slightly vary quantization by excluding notes from quantization that are very close to the next quantization value.

Q Window

"Q window" refers to the interval to the left and right of a grid point; events will be quantized within this range. No quantization will take place beyond this, and for this reason, the events outside of this window will remain at their position. The quantization area is dependent on the values of the parameters "Q snap" and "Q threshold".

Example: Grid: 4 max. Window: 4

- 100: The Q range covers the entire area between the grid points on the quantization grid. All events will be quantized
- 50: The Q range covers half of the quantization interval. Events with gaps of ¼ of the grid width left and right of the grid point (1/16 note values in this example) are quantized
- 0: no Q range -> quantization off

Select notes within the Q range

This button activates a display of the selected events that will be quantized for the currently set Q window size of the quantization range (Q range). The affected events are displayed with a red outline.

Click in an empty area in the MIDI editor to deselect.

The smaller the window size for the quantization area is, the fewer events will be included in quantization. The Q range and the graphical velocity display function help, e.g. to assign a lower velocity to all offbeat notes.

Swing

Set swinging, ternary playing with this value. This enables you to enter the division for an uneven/unaccented grid points.

- 50: "50-50/1:1" split. The unaccented eighths are exactly half way between the even eighth notes ("even", binary playing method)
- 67: "67-33/2:1" tuplets. The beat is split up into three counts, whereby the note is assigned 2 beats (67%) and the off beat note one count (33%)
- 75: "75-25/3:1" split. For example, a pointed eighth and a sixteenth is created from two eighth notes

Offset

The value range in this parameter stretches from -100 to +100. By changing the offset values, you move the whole quantization grid. Select a negative value for the offset, and the quantization grid will be moved by the corresponding value to the left, or forward in time. On the other hand, if you select a positive offset value, you will set the quantization grid by the corresponding value to the right, or backwards in time.

A value of -100 corresponds to a shift of half a grid length to the left; +100 corresponds to a shift of a half a grid length to the right.

Humanize

The "Humanize" parameter creates another variation option, i.e. notes are able to be assigned according to the randomization principle up to a specific interval to positions around the exact quantization value. The setting occurs in % of a 16th note. The value specified therefore determines the possible interval between the quantized notes and the exact quantization value.

Soft Q

This value sets the strength or "Soft Q" value of the quantization.

- "100" moves the event precisely to the quantize grid point,

- "50" shifts the event to the middle between the current position and the quantization grid point,
- "0" means no movement, i.e. quantization off

"Soft Q (quantize approximation)" (in the arranger menu "Object > Quantize > Extended") considers the current "Soft Q" value in the quantization options in contrast to "Note" and "Start" quantization.

In contrast, quantization commands "Quantize notes (start and length)" and "Start quantization" always proceed with 100% strength. This corresponds with a Soft Q setting of 100.

Note: Use the keyboard shortcut for the approximation of soft quantization and hard quantization. In this manner, you can always select between approximation soft quantization and harder quantization without having to switch quantization options every time.

Standard Quantization

Here you can choose what type of quantization is used as standard after pressing the "Quantize" button. You have the choice between:

Start Q: start quantization

Start and Length Q: start and length quantization

Length Q: length quantization

Soft Q: start quantize approximation

Apply standard Q: applies quantization according to the selection.

Note: If you give the "Soft Q" parameter a value other than 100, please take care that you also change the setting to the soft quantization mode. If the corresponding quantization settings should be considered, you need to set the "Soft Q start" command as the standard quantization. In contrast, quantization commands "Q start" and "Q start + length" always proceed with 100% strength (Soft Q).

Undo quantization/reset

With the help of the round arrow buttons, you can undo/redo the last quantization made. In this case, the counter-clockwise round arrow provides the "Undo quantization" function, while the clockwise arrow serves as the "Redo quantization" function.

Reset quantization: This function resets all quantization processes.

The "**Standard settings**" button provides the option of resetting the preset values:

Q threshold: 0

Q window: 100

Swing: 50

Offset: 0

Humanize: 10

Soft Q: 25

Step Recording with a Keyboard or Controller Keyboard

In the MIDI editor, so-called "Step recording" can also be executed using the computer keyboard or a MIDI keyboard. Activate the corresponding button first.

A marker shows the scope of the current octave for the following entries. Now you can enter MIDI notes step-by-step using the keyboard. Note length and step length can be specified by the length quantization values. The most important keyboard shortcut for entering MIDI notes using step recording:

Tab	One step forwards (set pause)
Shift + Tab	One step backwards
Ctrl + arrow up/arrow down	Entry octave upwards/downwards
CDEFGAB	Note entry in current octave
Shift	Enter chords

Different notes can be entered without the playback marker jumping if you hold down the "Shift" key. This enables chords to be played.

Notes can also be entered in "Step recording" mode by using the controller keyboard.

Cell Edit Mode

This mode may be opened/closed manually by clicking the "Cell" button beside the horizontal scroll bar. "Cell edit" mode provides an alternative view of the MIDI events:

- Notes are displayed as cells. In this case, the actual note lengths for the individual events are no longer considered. You may specify a unified display width of all events a lot more by setting the snap quantization and length quantization values in the MIDI editor's toolbar.
- The intensity of color of the individual notes increases with velocity strength – for larger velocity values, the event is displayed in dark blue, and events with little velocity are highlighted light blue.
- "Cell edit" mode provides a better overview for display of MIDI percussion instruments (see drum editor (view page 354)), since these often appear as short, explosive events. Display is limited to the most important information, note starting points, and velocity.

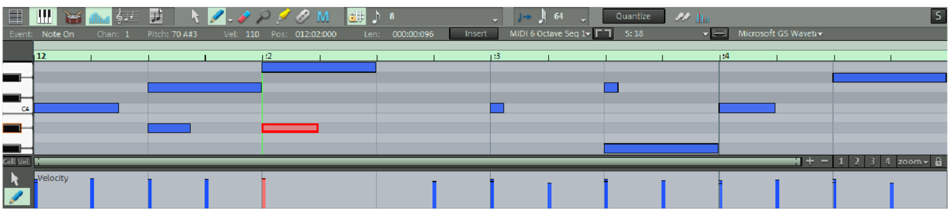
Velocity Mode

If you click the "Vel" button next to the "Cell" button, the view switches to "Velocity" mode.

In this case, the velocity strength is no longer displayed like before (with different color intensity of events). "Velocity" mode keeps the events the same color. The respective velocity may be recognized by the height of the event. The event's velocity may be changed by dragging the mouse to the upper edge of the respective event. In all mouse modes (except for "Delete"), the velocity can be changed by directly clicking without having to switch to the controller editor.

> **Note**: "Edit Shortcuts" in the MIDI editor menu enables the parameter "Event velocity" to be changed in the MIDI editor via a shortcut assigned by you. The functions "Velocity up" and "Velocity down" are available.

Matrix Editor (Piano Roll Editor)



For all MIDI objects that show no entry in the "Drum Map" field, you can open the Matrix Editor with a double click. The button is selected in the MIDI editor if the Matrix Editor is active.

The Matrix Editor is easy to recognize via the piano icon on the left side of the editor. The MIDI events displayed are arranged according to their respective pitch.

Note display

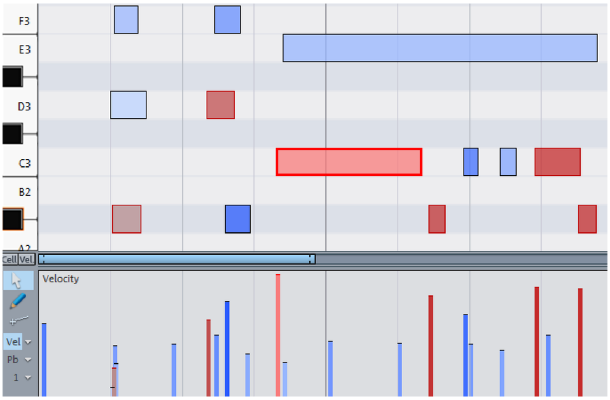
Notes which are not selected within the editor are displayed in blue. The intensity of the colour represents the velocity. The velocity increases while getting darker/stronger in color.

Selected notes are displayed in red. Here too, a more intense color symbolizes increased velocity

Current event: Selected events appear in bright red with a red border. The properties of the currently selected events are displayed in the edit fields above the Matrix Editor If an event is selected with the mouse, it turns into the current event.

Use Velocity colors (MIDI editor Options menu)

With Velocity Colors the velocity of events is displayed by different colors. The higher the velocity, the lighter the color.

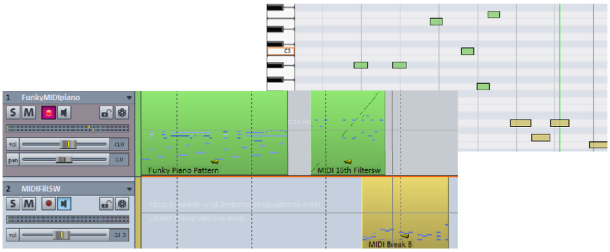


Velocity colors: Unselected note (blue), selected notes (red) and current events (red with border) in the Matrix Editor

You can also determine the colors of tracks or MIDI channels via the "Options" menu in the MIDI editor. If no previous track colors have been set, random colors will be assigned for track color display in the MIDI Editor.

Use Track Colors

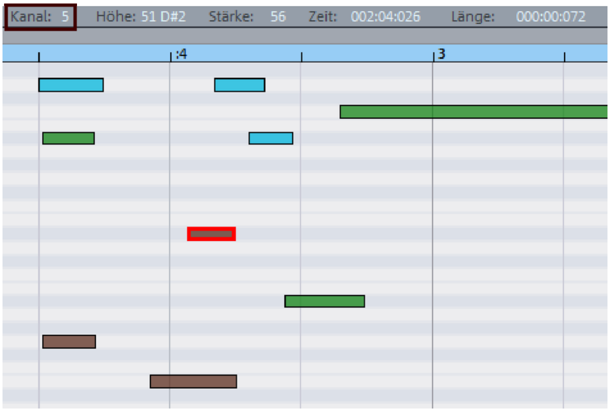
With Track colors the color of the events is set to the color of the corresponding tracks the MIDI objects are located in.



Track color display for MIDI events

Use MIDI channel colors

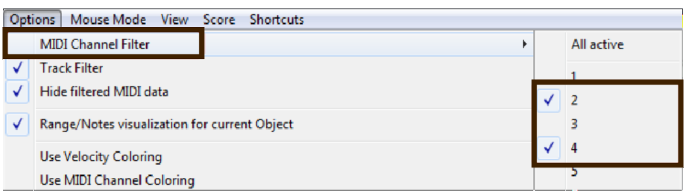
In the MIDI channel color display MIDI events are displayed in different colors, based on the MIDI channel settings.



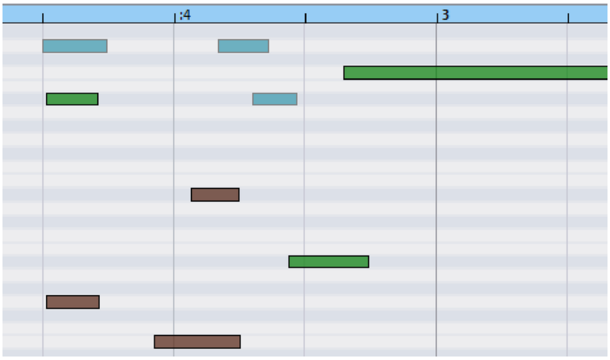
MIDI channel color display

MIDI Channel Filter: To get a better overview of the events of a MIDI object, you can filter specific events for the display.

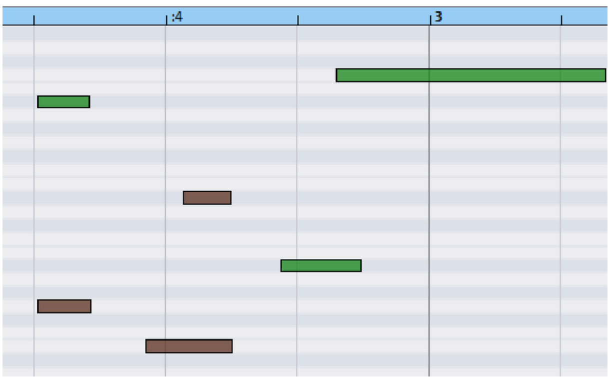
Example: The MIDI object features notes in the MIDI channels 1, 2, and 4. By selecting both MIDI channels in the MIDI editor menu "**Options -> MIDI channel filter**", you may make all notes in channels 2 and 4 available to editing tools.



All unselected notes in channel 1 will be filtered and will be displayed in gray/pale in the Matrix Editor and in the Event list.

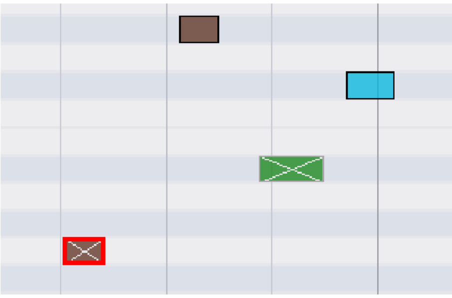


Events in the filtered channels may be hidden completely using the "Hide filtered MIDI data" in the "Options" menu.

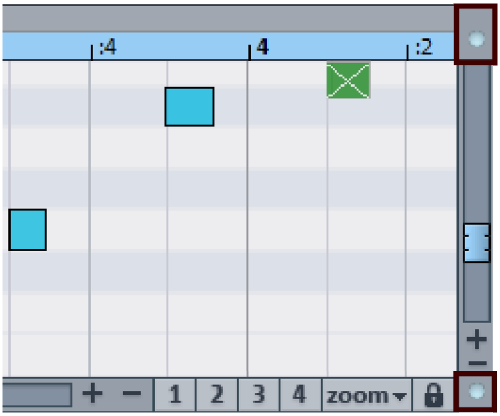


The Event list (view page 365) provides additional display filters that only function within the list.

Display of muted events: MIDI object settings (view page 326) (shortcut: "Ctrl + O") may be used to mute notes and filter other MIDI events. Muted events appear paler in the Matrix Editor and Event list.



Events above and below the current picture section: Two small red rectangles above and below the vertical scroll bar to the right-hand side of the MIDI editor screen show in red if there are notes outside the screen's display.



Matrix Editor: Special selection options

In order to directly select all notes of a certain pitch, double-click on a free section with this pitch in the Matrix Editor or in the keyboard. Hold down the "Alt" key to select notes with this pitch from this click position only.

Using the "Shift + double click" key combination on a free area, you can create a new event and simultaneously select all notes of this pitch.

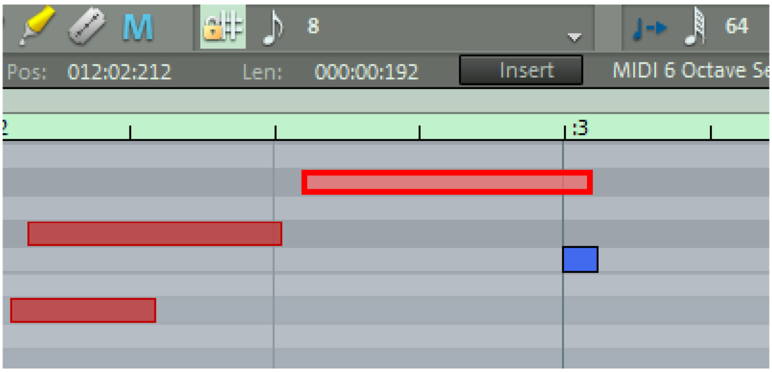
Note: If you additionally press "Ctrl" during any of these selection methods, the new selection is added to the current one (i. e. the previous selection is not canceled).

Matrix Editor: Edit events

If you move the mouse over an event the pointer will change depending on which event bar area the mouse is in. The following options are available:

- Change event start time: drag the event's front edge. The note end remains intact.
- Change event length: drag the event's back edge. The note start remains intact.
- Set fixed event length: for multi-selection, hold "Shift" and drag the currently referenced event longer or shorter. This function makes all event equal length.
- Scale event length relative: for multi-selection, hold "Ctrl" and drag the currently referenced event longer or shorter. The lengths of the other selected events change.
- Move events horizontally: move the mouse over an event whilst holding the "H" key. The mouse pointer will turn into a horizontal double arrow. Move the selected events horizontally in the timeline. The step size is defined by the set grid value.
- Move events vertically: move the mouse over an event whilst holding the "H" key. The mouse pointer will turn into a vertical double arrow. Move the selected events vertically in pitch.
- You can temporarily remove the snap and freely drag the event by holding "Alt".

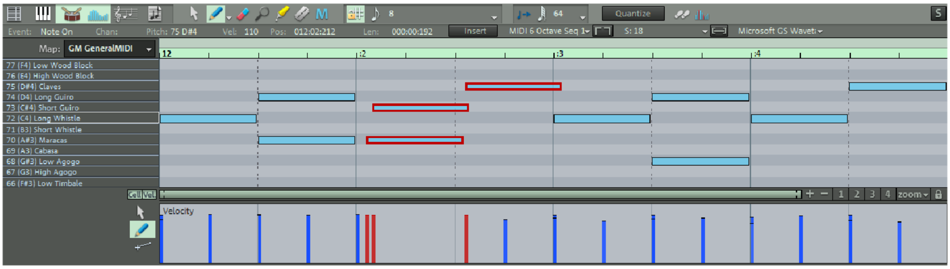
Relative snap for moving operations ("Options -> Relative movement in grid"): If this option is active, the interval between events to the next snap position remains for event moving. This makes preferred implementation of instrument groups easier to rearrange.



Drum Editor

After opening the MIDI editor (double click on the MIDI object), switch to the corresponding field

in the Drum Editor. Instead of the piano keys, a list of drum instruments is displayed.



> **Note:**If a drum map is active in the arranger track (recognizable by the word "map" in the MIDI area of the track editor (view page 95)), the drum editor is loaded automatically when the drum editor is loaded.

Drum editor track box: In the individual track boxes of the drum editor, you can individually adjust MIDI channel (K), grid (#), quantization length (L), display width in cell mode (< >), and the velocity scaling in percent (V) for each drum instrument.

Display width in cell mode: You can determine the display width of the notes of the current snap in percent with this value. The setting has no effect on playback. At "100" the displayed note completely fills the cell.

Velocity scaling: The velocity value of each note is multiplied by the V value/100 and adjusted to the MIDI velocity between 1 and 127. Scaling is audible, but is not visualized further.

When the drum editor opens cell edit mode (view page 348) activates by default. Here, moreover you can turn Velocity Mode (view page 349) on.

Mouse modes

The following modes are available for editing events, just like in the piano roll:

Selection mode (shortcut: 1): Use this mode to select drum events, move, or change the length of one or several events simultaneously.

Draw tool (shortcut: 2): This mode enables you to draw in drum events manually. If the grid is active, the events will be quantized immediately.

Drum tool (shortcut: 3): This mode also includes a draw function. However, the quantization length is also included while the events are drawn.

Pattern tool (shortcut: 4): This mode allows you to draw entire drum patterns (or melody patterns). If you want to create a new pattern, you have to select it first in "Selection" mode and press "Ctrl + N" simultaneously (or go to menu "Edit" in the "MIDI/drum editor -> Create pattern from selection"). If you have created a pattern, you can start drawing at any position. The lowest note in the pattern is then the pitch you are drawing.

Velocity tool (shortcut: 5): This mode allows you to mark events and change the velocity values of all selected events in relation to each other. Absolute values are entered when you hold the "Shift" key, i.e. all changed events receive the same velocity value.

Eraser (shortcut: 6): This tool deletes events with a single mouse click.

Zoom mode/magnifying glass (shortcut: 7): Draw a rectangle. The tool will zoom into the rectangle. The left mouse button is used to zoom in, the right one is used to zoom out.

Velocity

If this button is active, the bar height is used when displaying the velocity value of the note.

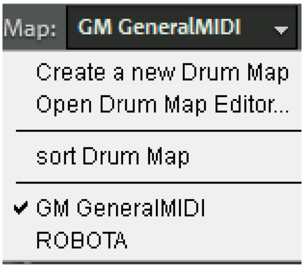
In all mouse modes (except for "Delete"), the velocity may be changed by clicking directly on a note without have to switch to the controller editor.

Drum Maps

A drum map assigns specific properties to the various parts of a MIDI drum kit. The key assignment for the pitch, the output note, the MIDI channel and the quantization are all set here.

The "General MIDI" map is used by default to distribute drum kits.

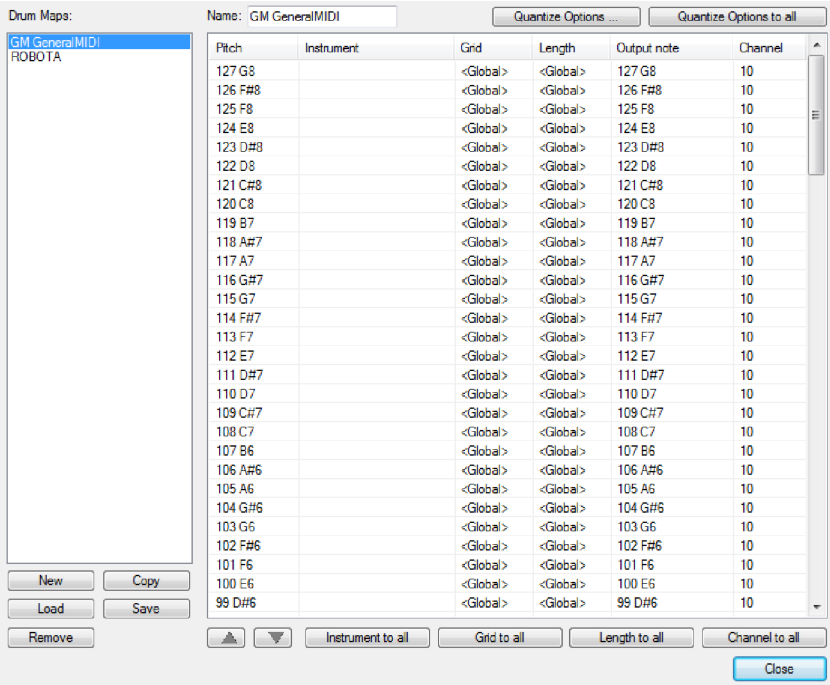
You can set the drum map in the drum editor by selecting the "map" field in the desired drum map. You can also open the drum map in the track editor under "MIDI".



It may be the case that your synthesizer, VSTi or drum machine uses a different mapping setup. If this is the case, you will hear a different sound than you expect when playing the drum events (e. g. a tom drum instead of a bass drum). Here we recommend creating a drum map that corresponds to the current playback device. To do this, select the command "Create new Drum Map" and assign the individual instruments to the corresponding pitch or key assignments of your MIDI keyboard. You can also specify the quantization settings and the MIDI channel. When the new drum map is saved it will appear in the selection menu.

Drum Map Editor

This editor enables you to route each played note to a different one and give them their own individual names. It's also possible to assign each individual instrument its own quantization and a new (MIDI) output channel.



Pitch: This parameter indicates the incoming MIDI note. The value cannot be changed, so the pitch always corresponds to the input note.

Instrument: Name each corresponding percussion instrument here.

Grid: Specify a grid for the start point of the drum events individually for each instrument. Leave the value set to "Global", and the globally set quantization value in the toolbar will be adopted.

Length: Specify a grid for each instrument for the length display of drum events. Leave the value set to "Global", and the globally set quantization value in the toolbar will be adopted.

Output note: Specify the note value each drum instrument or the incoming MIDI note in the "pitch" field" will be "mapped".

Channel: Specify a defined MIDI channel for each instrument. The value set here disables the channel setting in the MIDI track.

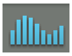
Note: In addition, you can set detailed quantization settings for each instrument in the drum map editor using the "Quantization options" button.

Quantization in the Drum Editor

You may also assign individual quantization values to drum instruments. To make the necessary settings, select the arrow next to the track name of the instrument you want to edit and open the quantization options (view page 339).

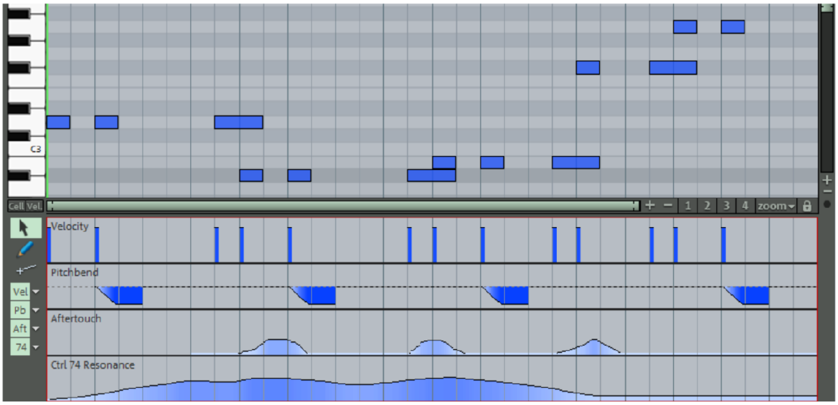
Controller Editor

The controller editor is located below the matrix or the drum editor; this may be hidden or shown via the keyboard shortcut "Alt + V" or the button for the same function.



Pull the dividing line beneath the horizontal scroll bar up to enlarge display of the controller editor.

The controller editor is used to display and edit up to 4 different MIDI controller curves simultaneously.

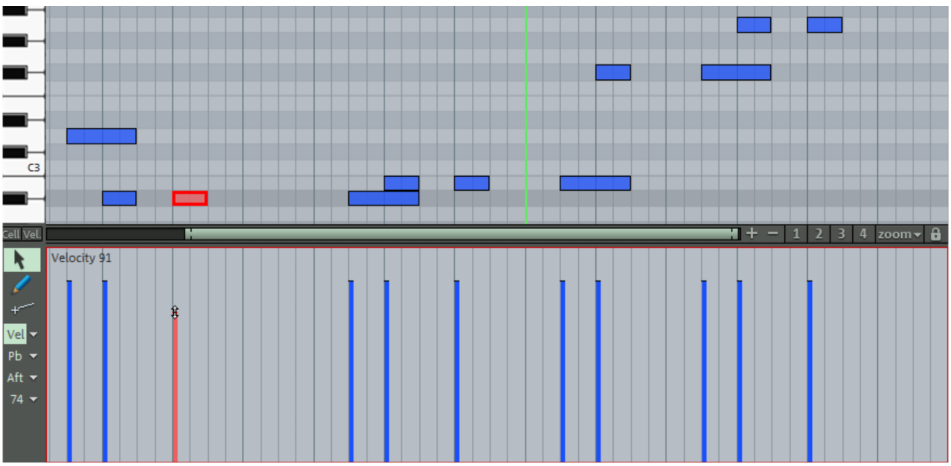


Note: To display more than one controller button, drag the dividing line between piano roll and controller editor up (the mouse pointer appears as a double arrow under the scroll bar for the piano roll). If you now click on one of the controller buttons, the corresponding controller line will be refreshed.

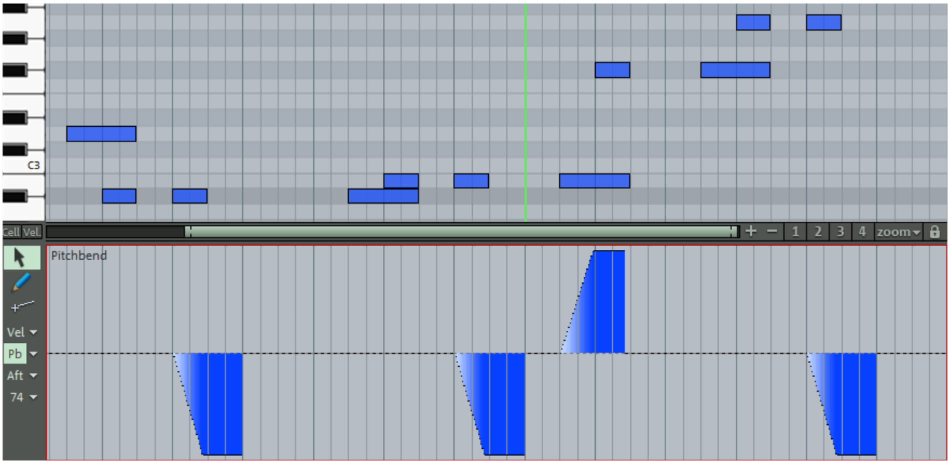
If you click on the arrow on the controller slot to select the displayed controllers, the following types can be selected:

- **Velocity**
- **Pitch bend** (bend the note pitch, just like with the pitch bend wheel of a keyboard)
- **Aftertouch** (pressure on the key after the first strike to control additional parameters, specified via the MIDI playback device)
- **Program change** (switch program for assigning preset sounds to the corresponding MIDI device)
- with a continuous **controller** numbered 0-127. The controller selection features the controller types available to the respective object; a star behind the name of the controller type indicates that a controller curve already exists for this controller type.

The velocity values are displayed directly in the controller editor as vertical bars under the corresponding notes. The height of each bar represents the corresponding note velocity strength. As velocity increased, the color intensity of the bars also increases. Selected events appear red in the controller editor.



The values of all other controllers will appear in the controller editor as ramps. In this case, the height of the ramps and their color intensity also represent the last defined value of an event. The length of the ramps reaches to the next varying event. Selected event ramps also appear red.



Controller Editor: Selecting, editing and drawing events

Combi tool:

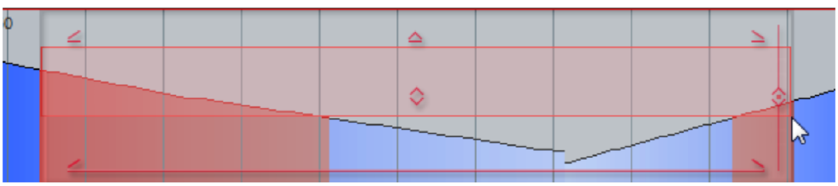
Keyboard shortcut:	Ctrl + 1

The combi tool enables you to select events, edit values for the selected events and draw straight lines or freestyle curves.

Selecting and editing events

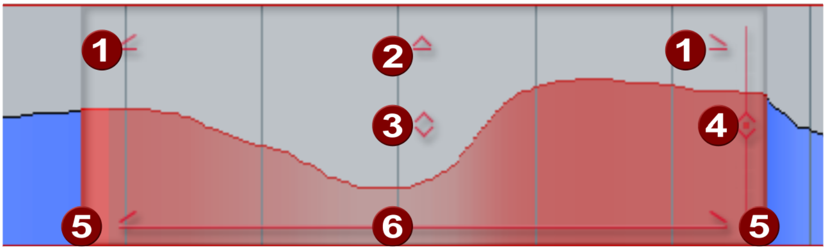
Individual events can be selected by clicking the respective bar. To modify the controller value, click and drag on the bar end.

To select multiple events click in the controller editor and drag to create a selection (lasso). The reference for the selection is the top edge of a bar. Since lasso selection is two-dimensional, it is only possible to select events in a specific value range.



Additional events can be added to the selection by holding down the Ctrl key.

The selected events can be grouped for editing using handles at the selection frame:



1. **Fade in and out:** Drag the handles to fade controller values in and out.
2. **Scale values:** Drag to handles to increase or decrease values in relation to each other.
3. **Increase/decrease values:** Drag the handles to increase or decrease all values to the same value.
4. **Increase/align differences in value:** Click a point on the vertical line and the differences in value on either side of the point will be increased or decreased. If you drag the handles right down to the bottom, this creates a straight line at the height of the clicked value and the "amplitude" of an existing curve will be increased in the other direction.
5. **Stretch/compress duration**: Drag at the edges of the selection for temporal stretching or compression of the curve.
6. **Move selected:** Drag down at the line to move the selected curve.

To delete selected events in the controller editor, press the "Del" key.

Other editing options with the combi tool:

Alt + drag (with the pencil): This activates freehand drawing, and the mouse pointer becomes the pencil. Detailed envelopes, ramps, and grades can be created by dragging within the respective controller range.

Shift + drag (with the crosshairs): This activates line drawing and the mouse pointer changes to crosshair. Create envelopes, ramps and grades, or set all of the events to the same value by dragging horizontally.

Draw freehand

Shortcut key: Ctrl + 2

This activates freehand drawing; the mouse pointer becomes a pencil. Detailed envelopes, ramps, and grades can be created by dragging within the respective controller range. Correct curves during drawing by dragging backwards. Any existing (multi) selection is ignored. Create crescendo or decrescendo using velocity curves, for example.

Single-clicking creates a new ramp which reaches up to the next controller event.

Shift + Drag with the crosshair to draw a line.

Note: When editing the velocity, no new notes are generated; existing velocity values are only modified by dragging and clicking the events.

Draw lines

Shortcut key: Ctrl + 3

This activates line drawing; the mouse pointer changes to crosshair. Create envelopes, ramps, and grades by dragging horizontally.

Single-clicking creates a new ramp which reaches up to the next controller event.

Note: When editing the velocity, no new notes are generated; existing velocity values are only modified by dragging and clicking the events.

Controller Editor Tips

Vertical zoom

To make it easier to edit very small curves (e.g. pitch bend), you can use Ctrl+mouse wheel to zoom vertically into the controller curve.

Copying controller events

You can copy selected events using "Ctrl + C" and then paste them at any play cursor position in the corresponding controller editor using "Ctrl + V". This also works from one controller to another. With Ctrl + A you can select all events of a controller.

Display filter function for velocity values

At points in the arrangement where there is polyphony the velocity bars for the notes are positioned on top of each other and it can be difficult to select the right note. To edit only notes of a certain pitch, e.g. all C1 notes in the controller editor, click the corresponding key on the keyboard. The key and the background of the selected pitch are highlighted. Now only notes of this pitch are displayed in the controller editor. Repeated clicking of the same key removes the selection.

You can also select several pitches for the velocity visualization filter by clicking on the desired buttons while holding down cmd or Shift.

A further possibility for selectively editing velocity bars located on top of each other is based on the fact that only the bar of the currently selected event can be edited using the mouse. First click on an event in the matrix editor or controller editor. Now navigate to the desired note with the "Arrow left/Arrow right" buttons and change the controller value by clicking the top third of the selected bar.

Quantize Controller Events

MIDI controller events can be quantized and thinned out. To do this select the "MIDI functions" menu and then the "Quantize/thin out controller" command. Quantization occurs according to the quantization settings.

> Detailed information about quantization is provided in "MIDI editors -> Quantization".

Smooth controller events

It is also possible to smooth controller event curves. To do this, select "Smooth controller" from the "MIDI Functions" menu. This adds interpolated intermediate values between existing controller values, which creates a smoother curve gradient and avoids jumps between values.

Event list

Open event list: The MIDI editor has an integrated display of all events in list format. This List Editor can be opened with the button above the keyboard or by using the "Alt + L" keyboard shortcut.

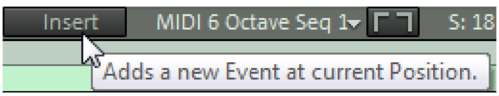
When the event list is opened and activated for editing, it has a narrow red border. This is to make clear that certain functions, e.g. select next/previous event (arrow keys) or the "Select all" (Ctrl + A) command, refer only to the List.

Working with the list editor: The list editor does not only display note events, but also percussion instruments (in drum editor mode), MIDI controllers, and SysEx messages. These controllers and messages may be hidden and even filtered by checking the appropriate "mute" box.

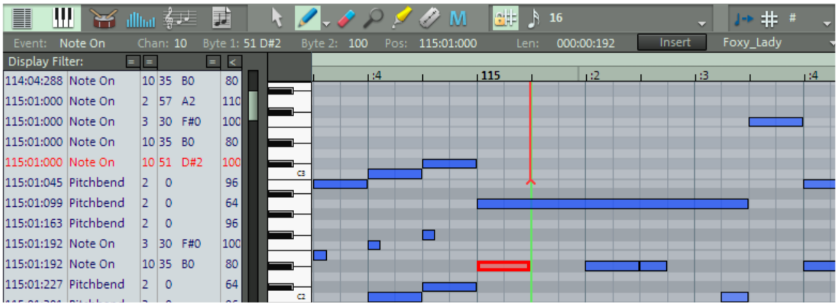
Double clicking on a SysEx entry in the list opens a simple editor for viewing and editing the SysEx information.

A note on and a note off event (or note on with velocity 0) belong to a note. These are always selected and edited in pairs. You can show and hide note off events by ticking the corresponding checkbox beneath the Editor.

You can create new events in event list directly at the playback marker position using the "Insert" button.



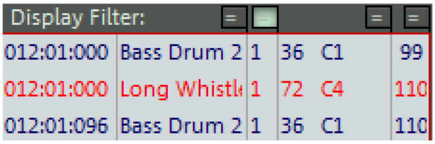
The values of the previously selected notes for the edit fields channel, byte1 (Pitch), byte2 (velocity), time and length are used at that.



If you would like to edit only certain events specifically, the event list offers various display filters. These are small check boxes above the event list columns. If you click a box, it will change to blue and the respective display filter is active.

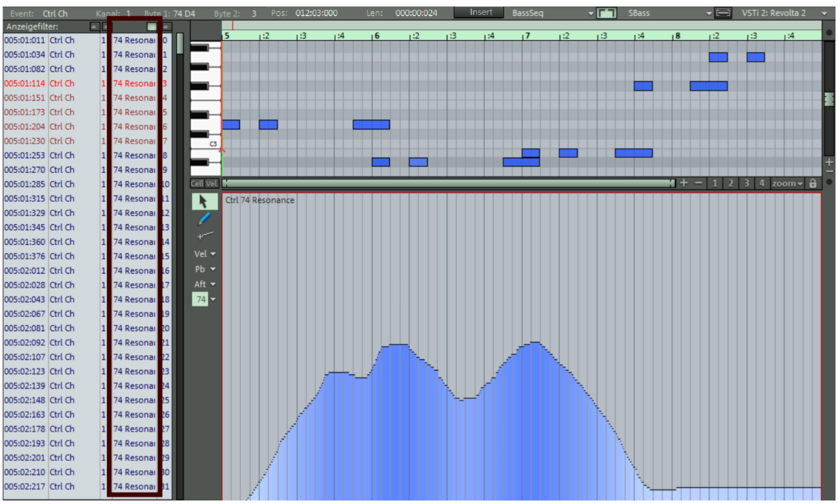
The display filter provides information for the editing fields MIDI event, MIDI channel, pitch and velocity.

Example: Select a note and click a display filter of the pitch column to only display events of this pitch. All other events will be faded out:



Display Filter:					
012:01:000	Bass Drum 2	1	36	C1	99
012:01:000	Long Whistl	1	72	C4	110
012:01:096	Bass Drum 2	1	36	C1	110

Display filters can be combined with one another. For example, all control change events of type 10 (pan) to MIDI channel 1 may be filtered for display. Then, you can select all matches will "Select All" (Ctrl + A) and edit together in the editing field or delete with the "Del" key:

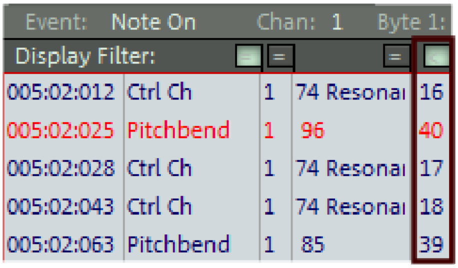


Advanced filter functions in the list editor

You can access the following filter functions by right clicking on the display filter buttons:

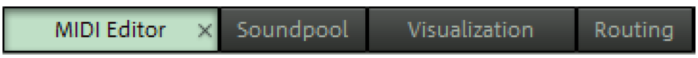
=	equal
!=	unequal
>	bigger or equal
<	smaller or equal

The filter functions listed above relate to the selected event kinds like "Note On" or "Ctrl Ch". For example, this function filters out all events with a value of 40 or lower:



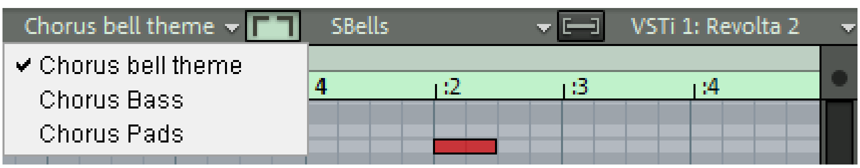
Multi object editing (MO-Editing)

You can also load several MIDI objects simultaneously in the MIDI editor. All MIDI objects selected in the arranger are added to the MIDI editor by clicking the "MIDI editor" button in the Docker (view page 90).



In an already-opened MIDI Editor you can include additional MIDI objects for Multi Object editing by clicking in the Arranger while holding down the Shift key.

The MIDI editor (and in multi object editing) always shows the current MIDI object and the current arranger track from which this object originates. In the field to the right you'll see the MIDI out of your sound card or the activated VST plug-in for this track.



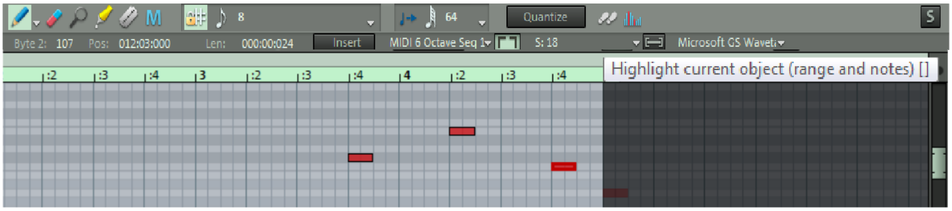
New MIDI events will always be inserted into the current object (i.e. the currently displayed object). The current object is also highlighted in the score editor.

If you click on the arrow next to the object/ track display, you will see all MIDI objects/MIDI tracks in MO Editing mode listed beneath one another. Here the current object or current track has a tick.

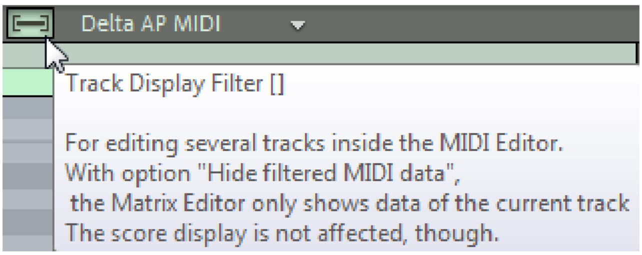
When notes or MIDI events are selected, the associated object becomes the active object. The current Note line will be automatically converted in the score editor.

You can also copy and insert MIDI notes between multiple objects.

The current object range can be highlighted by clicking the button behind the object selection field. Areas outside the current object will be grayed out and the notes of other objects will be paler but still selectable.



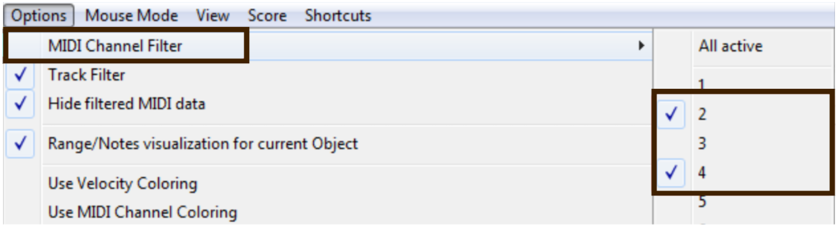
In addition, a display filter for the current track can be activated while in MO editing mode. Events on other tracks will then be grayed out. This also reduces the display in the controller editor to the velocity values for the current track. If the option "Hide filtered events" in the "Options" menu is also activated, MIDI events that are not current will be made invisible.



Note: When representing notes, the note system of the current track will be highlighted using blue note lines.

Filter by MIDI channel

With the "Options menu" > "MIDI Channel Filter" in the MIDI Editor you can select the MIDI channels that should be displayed.



Th disable the MIDI Channel Filter, just select "All active".

Explanations about the colorig of MIDI events can be found in "Matrix Editor“ > "Note display (view page 350)“.

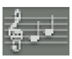
Score Editor

The score editor displays the MIDI data of a MIDI object as notes in real time and also offers all MIDI editing options in the note display. If you move or extend MIDI data, these changes are immediately reflected in the score. If you insert a new note in the score editor, a corresponding "MIDI note on" event is created immediately.

Each track may contain a maximum of 48 staves. During multi-object editing across multiple tracks, the scores from the systems of each track are compiled together. The staves of a track may be used as an instrument or instrument group within a score. The entire score may be revealed via the multi-object editing feature and by showing all MIDI tracks in the score editor simultaneously. A vocal excerpt can be obtained by showing the track of the desired instrument only or an instrument group in the score editor.

Open Score Editor

The score editor is integrated into the MIDI editor window. If the MIDI editor is opened, you can activate the linear score view by pressing the "Score editor (linear)" button.



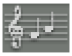
Score editor (linear) button

Score Editor Modes

Samplitude offers two alternatives for viewing the score: linear display and page mode.

The linear note display may be combined with the matrix display. This offers ideal MIDI editing options, since the detail in depth of the matrix editor and overview may be supplemented by several score systems. You can select the score in the score sheet and perform fine adjustments in the piano roll. Selection and zooms in all editor views (linear score, piano roll, event list, velocity editor) are always synchronized.

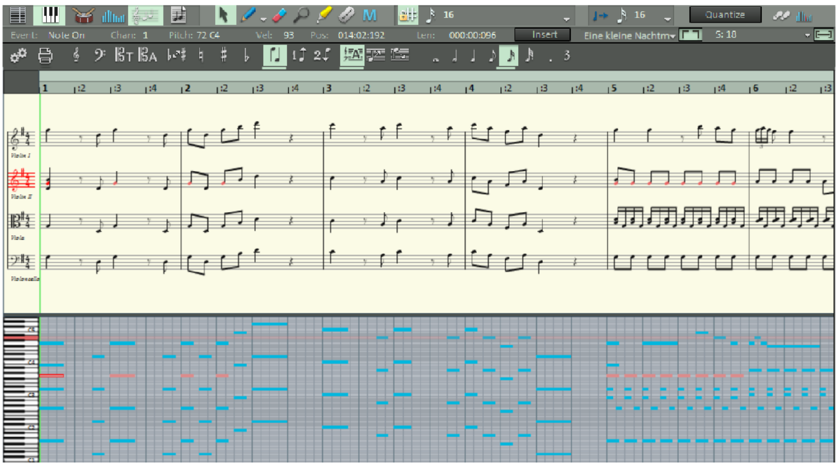
Note editor modes - Linear view



Score editor (linear)

In the linear view, not only the Score editor, but also the matrix and velocity display are available. For detailed editing of the MIDI data in the score, select the linear view. All parameters such as pitch, note length, and velocity are displayed.

Note: Only the notes of the currently active system are displayed in the velocity editor if notes and velocity are parallel. The red bar mark indicates the currently active system.



The linear height of the range of the note view may also be modified. To do so move the line between Score and Matrix editor with the mouse.

The horizontal section can be set above the scroll bar, parallel to the matrix view of the MIDI notes.

In "Linear" mode, details may no longer be displayed correctly if you zoom out horizontally, since the note symbols overlap. It may be necessary to reduce the notation symbol size with the "-" button on the right side. Despite these details being displayed incorrectly, zooming out can still provide you with a good overview, particularly for large scores. Samplitude can adjust the notation symbol sizes automatically at the selected section. To do this, please select "Automatic zoom" in the "Score" menu.

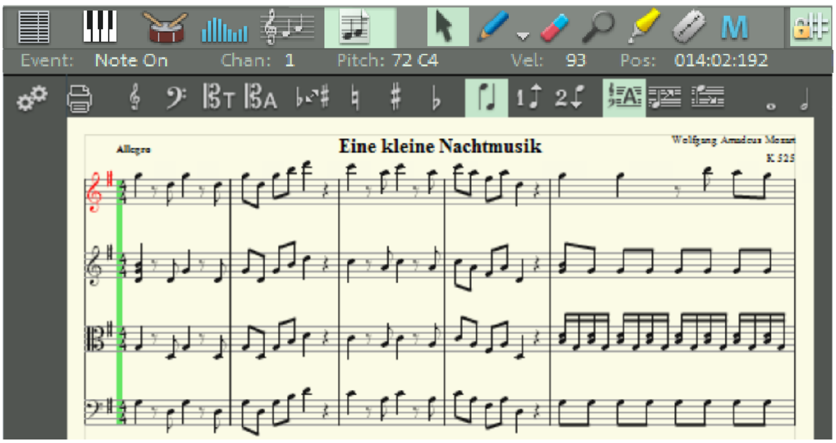
Score editor modes - page view

The side mode displays the notes just like on a musical score, which also serves as a print preview. This enables you to arrange or delete notes or voices. The design of the score sheet can be used for editing longer passages. The stave break allows you to display considerably more bars than with the linear display.

Score editor - Page view

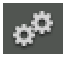
Scrolling in full view mode: Use the vertical scroll bar to scroll from page to page.

Page mode as print preview: You should first change the page format (in the Score Properties dialog) to the paper format of your printer before setting up the page for printing as the display depends on the page format. You will then see your score exactly as it will be printed.



Score sheet

The automatic conversion of MIDI data into score material is usually sufficient for editing MIDI data; in this case, efficiency is required instead of a perfect, ready-to-print display. However, it may still be necessary to adjust some of the display options for the musical material during editing. These display options are provided in the "MIDI score properties" dialog, which you can open by clicking the corresponding button.



Score settings: Open the score settings dialog for notation system and page properties

Presets are also located here. These are preprogrammed standard settings for certain instruments or instrumentation such as strings, piano, or orchestra. Selecting a preset changes the score display and its legibility:

Before...

...and after the selection of the "Piano" preset

Further explanations of the notation system (view page 378) are provided below.

Editing MIDI data in the score sheet

Select notes

As usual, you can select notes by clicking them. A group of notes (a chord) may be selected by dragging a frame over the notes with the mouse button. Select multiple single notes by pressing the "Ctrl" key.

Note parameters

Parameter pitch, velocity, and length can be changed for one or several selected notes. If one or several notes are selected, the data for the current note will appear in the info bar above the score view. Changing a parameter may have a comparable effect on all selected notes, just like in the matrix editor.

Note: In "Page" mode, notes may not be drawn with the pencil or moved or copied with the mouse. Use the corresponding menu commands and keyboard shortcuts. This limitation does not apply in linear display mode.

Move and transpose

To move notes, first select them, and then drag them to the desired position. The infobox can help you keep track of the pitch or position. The step size when moving is determined by the selected quantization raster in the MIDI editor.

Copy

Select the desired notes and copy them by holding down the "Ctrl" key and dragging the mouse. Alternatively, you can also use the copy function in the "Edit" menu.

Insert new notes

You can also insert new notes in the score editor with the pen. Click the desired position with the pen, hold down the mouse key and correct the position and pitch as required. If you let go of the mouse key, Samplitude will add a new note featuring the same length as the selected length quantization value.

You may only insert new notes into the active stave. For instance, to insert a note in the lower part of a piano system, click the lower staff in the system on the left first. Only notes that correspond to the current pitch are inserted. Non-scaled notes or chromatic intermediate steps are skipped. If you enter conventional music material with the mouse, diatonic insert mode increases the chances of hitting the correct note. If a new prefix is going to be added to the note, you can move the note chromatically with the arrow keys. This way, an inserted F in C Major may be transformed into an F# by pressing the "Page up" key.

Delete notes

Delete notes by

- selecting them and pressing the "Del" key or
- clicking the eraser tool or
- by clicking them with the right mouse button

The menu command "Score -> Selected notes: hide/show in score" enables the selected notes to be removed from the score, without influencing MIDI playback. Notes hidden in the scale are indicated in the matrix by a diagonal line. This function is practical for making trills easier to manipulated, or in order to to remove "control notes" from the score when switching between playing styles.

Insert notation symbols

Clef symbols may be inserted at the current cursor position by pressing the corresponding clef symbol button in the active system.

Delete notation symbol

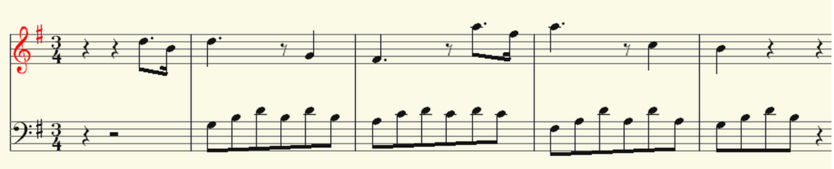
Manually added notation symbols such as clefs and signatures may not be selected, since these are meta information for the notation display and no MIDI events have been assigned to them. Nevertheless, these may be deleted by clicking them with the eraser (or the right mouse button).

Adjusting and optimizing the score

Samplitude automatically generates a notation display from the MIDI events contained in the MIDI object. This is always correct with regard to pitch and position. However, this does not mean that the notation can be optimally read, since displaying note lengths also plays an important role in this context. In this case, the notation allows flexible interpretation, so the user usually has to intervene. This is a negative example of a poorly readable translation of a piano passage in notation:

Automatic notation with unsuitable display settings...

This representation may be correct; however, it is illegible and somewhat confusing. Why is that? The MIDI events contain very precise information about the start of a note and its length and pitch, which has to be taken into account during playback. It may influence the groove of a song if the notes are always slightly shorter than sixteenth notes. If this were to be displayed correctly in the notation, the score would be unreadable as in the example above. The MIDI events also do not contain information on whether the gap between two notes is a real pause, its harmonic correlations (pitch), and the characteristics of the dynamic sequences. This is why automatic processing of notation always differs from what would be ideal. Samplitude includes a number of automatically and manually controllable functions for making it easier to read the notation.The reworked version of the above score excerpt illustration shows how big the difference can be:



...and editing is finished.

Note allocation in multiple systems

The term "**System**" refers to an individual line in a staff as well as all staves of a score. In cases where it is important to be able to differentiate a score and a note line from one another, we use the term "System" for the score and "Staff" for a single system.

Mostly, what is meant by "system" is the result of the context. For example, a two-handed piano piece makes reference to the "upper" and "lower" systems.

Samplitude provides multiple systems, e.g. for piano notation or entire scores that are comprised of up to 48 individual systems per track.

To assign notes manually to a system, click the "Assign to upper system" button

,

to move the selected notes one line higher on a staff, and correspondingly on "Assign to lower system"

,

to move the selected notes a line lower. This results in the note being connected to the line (independently of the MIDI channel or pitch). This manually set allocation can be undone by clicking on

 Assign notes automatically

to ungroup them.

> **Note**: You can only move notes within the staff of the corresponding track during "Multi-object editing".

When transcribing a MIDI piano recording, splitting the notes into a two-line piano system using the split point is recommended. This indicates that notes above the split point pitch belong in the upper system and that the other notes belong below it. The positions where individual notes are placed incorrectly can be easily corrected by assigning the notes manually by clicking the desired system.

Automatic assignment of notes to a specific staff is flexible. Either the MIDI channel of the note event, the pitch, or even a combination of the two can make up the criteria. This permits simpler and faster distribution of MIDI notes in the staff.

Example: For some standard MIDI files, the notes of a lower system of a particular piano piece may have a different MIDI channel than that of the notes of the upper system. Assume the notes for the right hand are on channel 1 and the notes for the left hand are on channel 2. In this case, set up two staves for the system in the note system settings. The easiest way to do this is with the "Piano" preset. For the first staff, set the MIDI channel allocation to "Ch. 1" in the "Channel" selection box and "Ch. 2" for the second staff (see MIDI score settings dialog (view page 378)).

The rules for allocation are as follows: If the "Automatic system allocation" option (preset) is set for the note, the note lines will be played until the MIDI channel corresponds with it and the pitch is over or equal to the split point.

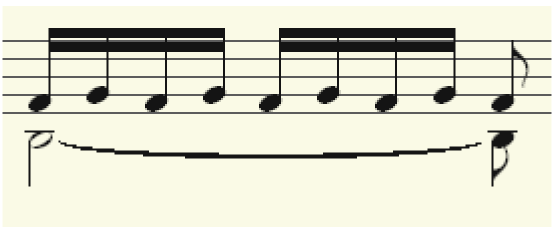
Note: Some notes may not show at all; that's because they haven't been assigned to a system yet.

Multi-voice notation

Up to two independent voices may be annotated for each staff. The voices differ in the direction of the note stems: the first voice is always with the stem pointing upwards, the second with the stem pointing downwards. Pauses are displayed individually for each voice.

Multi-voice notation can simplify the score considerably and enable multiple instruments or parts to be displayed in one staff together.

Mono-voice display



Multi-voice display

You can set the voice by selecting the notes and clicking:

"1. Assign voice (stem direction up)"

and

"2. Assign voice (stem direction down)"

. This sets the stem direction of the notes and therefore the voice assignment itself.

The set voice assignment can be ungrouped again by selecting

 automatic voice assignment

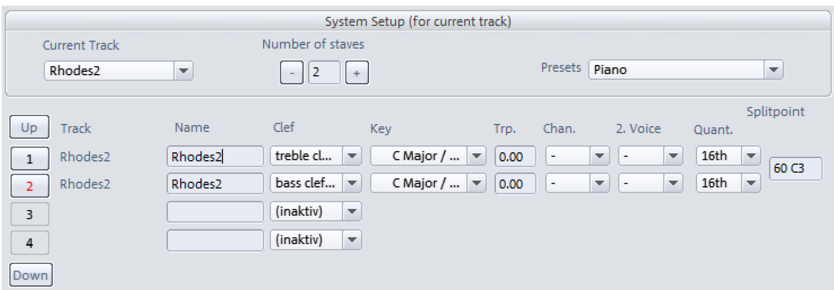
to ungroup them.

Tip: You'll find the corresponding commands in the MIDI "Score" menu; keyboard shortcuts may also be assigned to these.

For automatic voice allocation, the note's MIDI channel has to be analyzed.

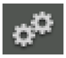
To do this, set up a MIDI channel for the second voice in the "Note system properties" dialog. This can be set individually for each staff. All notes of a system featuring a MIDI channel that has not been assigned to a second voice are added to the first voice.

Voice allocation via the MIDI channel can be practical for displaying standard MIDI files with multi-voice piano pieces, e.g. if the right hand voices are set to the first and second MIDI channels and the left hand voices are set to MIDI channels three and four. Apply the following settings:



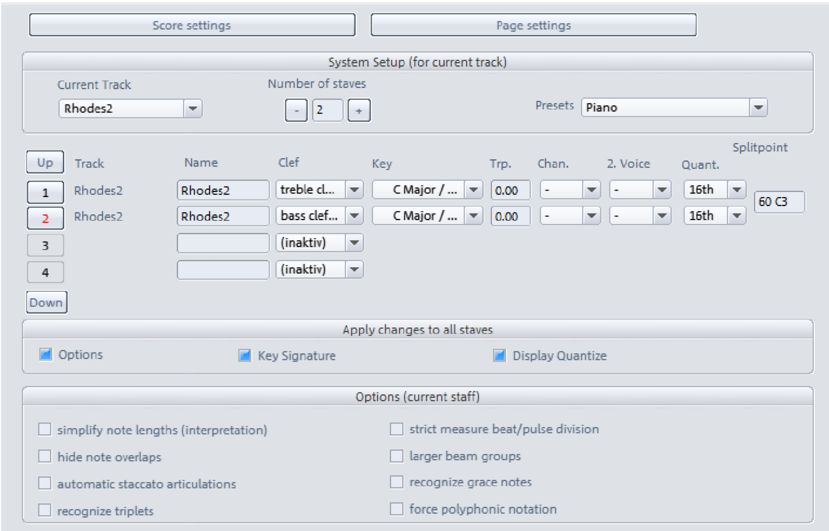
Note: If there is no MIDI channel selected for voice allocation, only one voice will be annotated (if manual voice allocation does not occur).

MIDI score settings



The score settings dialog opens the notation system and page format properties.

Apply the note system and page format settings here. None of the settings have any influence on the MIDI data itself, but rather only on the display of the score.



Note: The affected note system settings are always applied to all MIDI objects situated on the current track. The page format settings apply to the entire VIP project.

Note system settings

All templates for the system, the display parameters, and options are available on this dialog page. Specify the key, clef, and much more here.

Samplitude provides up to 48 staves on one track for MIDI data. Key, clef, display transposition (for transposing instruments like saxophone, etc.), and display quantization may be individually set for each staff.

An instrument's prefix ("Name") may also be set, plus the MIDI channel for automatic system/voice assignment.

Similarly, the split point provides automatic system allocation. Notes above the split point are added to the upper system, while those under the split point are added to the system beneath it (as long as the MIDI channel matches).

The active staff is recognizable via the index marked in red. The list of 16 staves can be scrolled vertically using the arrow buttons (up/down). The active staff's display options are displayed in the lower range.

Optionally, all changes to settings, display options, key, and display quantization are always transferred equally to all staves.

During multi-object editing across multiple tracks, the scores from the systems of each individual track are compiled together. A partial system of a track can be composed of multiple staves (e.g. 2 lines for a "grand staff" in piano notation). The system lines within a track will be grouped with struck-through beat lines.

The staves of a track may be used as an instrument or instrument group within a score. For this reason, consider early on how you would like to distribute multi-voiced pieces to multiple MIDI tracks to obtain a sensible score display. The entire score may be obtained by using the multi-object editing feature and showing all MIDI tracks in the score editor simultaneously. A vocal extract may be produced by showing the track of the desired instrument only or an instrument group in the score editor.

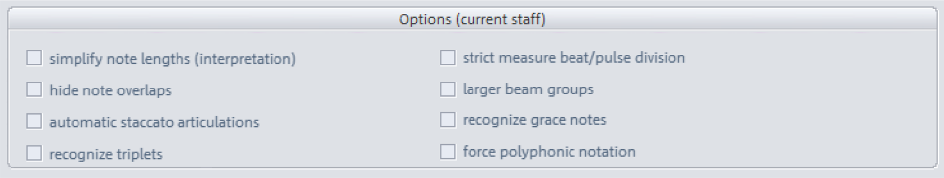
Display quantization

Using the display quantization ("Quant." column), you can set the rhythmic resolution of the note display independent of the actual quantization. This enables you to display a freely imported and unquantized track in sixteenth notes.

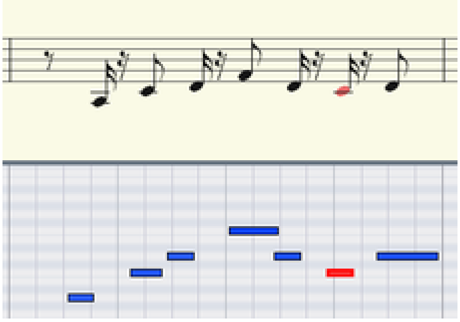
Set "Quant." to the lowest value in the sequence of the note value. For a sequence that contains sixteenth notes as the lowest value, select the setting "16th", not "64th" notes. A quantization display that is too fine may make the display illegible.

The display quantization does not have any influence on note playback, but rather adapts the note display to a raster. The actual recording (the MIDI files) is not changed, unlike the quantization function in the MIDI editor.

Interpretation Options



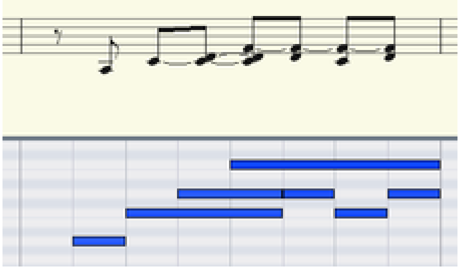
None of the options have any influence on the MIDI files or playback. An adjustment only occurs with regard to the note display. **Simplify note lengths (interpretation):** Displays pauses and slurred notes in such a way that the score is as legible as possible without influencing playback.



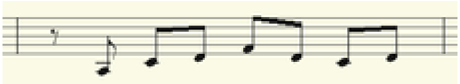
Display without active interpretation options; Display quantization is set to one sixteenth.

Display with option "Simplify Note Lengths (Interpretation)"

Hide Note Overlaps: With this option you can get rid of any overlapping of sequential notes which, for example, arises as a result of playing legato:



Original



Display with "Hide Note Overlaps" option

Automatic Staccato Articulations: A staccato symbol is added to a note whose value is considerably longer than the displayed MIDI note.

Display with additional option "Automatic Staccato Articulations"

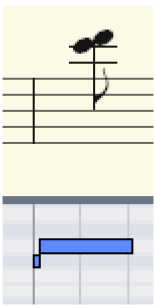
Recognize Triplets: Activate this option if triplets are present in the MIDI object.

> **Note**: The display quantization value always has to be one step finer than the smallest discernible triplet value. For example, to recognize one eighth triplets, display quantization has to be set to at least one sixteenth (or to 1/64 for 1/32 triplets).

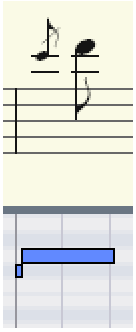
Strict Measure Beat/Pulse Division: This option ensures that there are no note or pause values longer than a beat subgroup (pulse). In some cases longer notes are displayed as multiple slurred notes. This can simplify the legibility of the score.

Compile Bar Groups: Bar groups are partially compiled across beat subgroups. This can also simplify the legibility of the score.

Recognize Grace Notes: Note values which are much shorter than the display quantization value are annotated as grace notes as long as a base note is present.



Original



Display with "Recognize Grace Notes" option

Force Polyphonic Notation: The display of the notation always appears with polyphony.

Notation symbols

Clef

There are four clefs available in Samplitude: violin, bass, tenor, and alto clef. Samplitude differentiates between base clef and clef change. The base clef may be set up for every system in "Score settings" (and applies to all MIDI objects on the current track). A clef that you enter via the toolbar will be interpreted as a key change. Key changes are possible any number of times in a song and also within beats.

To add a clef, activate the staff where the key should be inserted by clicking the staff signature to the left.

Position the playback marker at the desired insert location and click the clef you want in the toolbar. The clef will be entered musically rather than graphically.

Example: For a key change at position 10:01:000 (the beginning of bar ten), the clef symbol will be displayed at the end of bar nine just as it corresponds with the notation rules.

Key changes may also be deleted by clicking the clef symbol with the eraser (or the right mouse button).

Beat signature

The metric and beat signature symbols are created automatically from the tempo markers of the VIP project.

Time signature changes are possible at full beat borders. Create a beat counter marker with the new signature (e. g. 6/8) at the desired position via "Edit > Tempo -> Tempo/Beat marker (view page 395)". If there are no beat count measure changes, setting the time signature of the piece (e. g. 3/4) in the transport control is sufficient.

Prefix

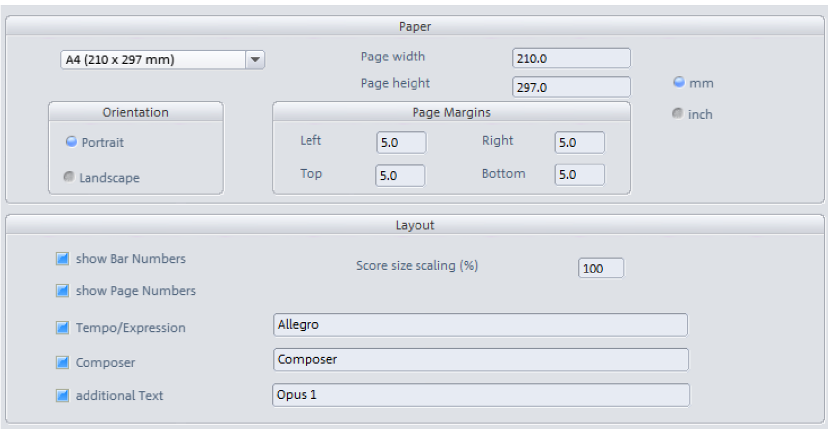
 Enharmonic change

Samplitude sets the sharps and flats according to the clef selected. It is often the case, however, that enharmonic changes can considerably optimize the legibility of certain passages. In this case, you can work manually. To change one or more selected notes enharmonically, click the corresponding button. The function transforms flats into sharps and vice versa.

Page format settings

Page format settings may be accessed via "Score -> Score settings" in the MIDI menu. Click the "Page settings" button.

You may select the paper format independent of the printer settings so that it always has the same note layout, regardless of where you are working and independent of the printers installed in Windows.



Samplitude automatically creates the layout of the score for optimal division of staves and systems on the page. You only need to specify the page size, orientation, and page borders.

Scale score size (%): Scales the size of the note symbols for the printout. However, the score display influences the position where a line or page break occurs.

- Set a value smaller than 100% to fit more beats/systems onto one piece of paper.
- Set higher values than 100% to keep larger note symbols on the printout.

You can also make entries for tempo notation, composer, and an extra text field. Similarly, you can choose which layout elements should appear on the page (check boxes for beat numbers, page numbers, and texts).

Print notes

Activate the print process in the "Score -> Print score" menu or via the "Print" button. The print dialog window will appear which has been adjusted to your printer or printer drivers. Depending on the printer, there are various options like the selection of pages you wish to print and the number of copies. Make sure to set up the same paper format for the printer as for the page mode, otherwise the printout may be scaled and the page ratio changed.

The following items won't be printed; they are only visible on the monitor:

- The lines which highlight the page borders on the monitor
- Mouse pointer
- The color display of the currently selected notes and playback areas.

Note: When printing a file (e.g. as a PDF file with a special printer driver), note that you have to activate the option "Save fonts in document" in the printer driver so that the notation symbols will be displayed correctly in the document.

Score editor – tips

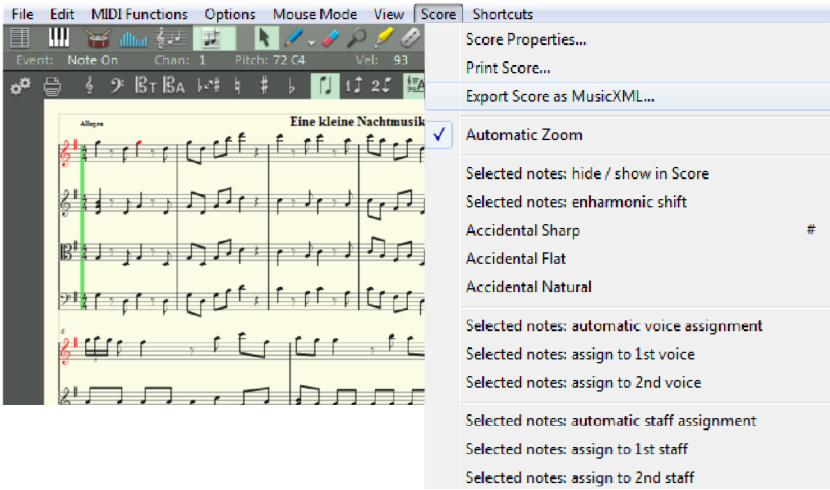
In complex arrangements, we recommend using two separate versions of a song for sequencing and for note print previews. This way, MIDI events for the note display may be changed in any way without influencing the playback. You may have to increase or decrease the length of a note in order to display the correct note value or the correct pause.

It may also be advantageous to remove trills, grace notes, and other ornaments in order to be able to print the score in a more simplified form. This manipulations, however, do affect playback. Compromises are not necessary when working with separate versions for sequencing and note print preview.

Score music XML export

The score may also be exported as a MusicXML file and then edited further in notation applications. Finale, Sibelius, Forte, plus many other MusicXML files can import music XML files. Converters are also available for transforming MusicXMLs into other formats. You can find out more at www.musicxml.org.

During the MusicXML export, all properties of the score (except for the page layout), like staves, notes and pauses, bar groups, keys, polyphonic voices, lead signs, enharmonic change, etc. are transferred. However, page layout, line breaks, and the distance between systems are not transferred. These may also be a component of MusicXML, but since these parameters in Samplitude may only be partially edited, they will be removed from MusicXML export.



Not every staff editing application processes all MusicXML elements correctly, so it may become necessary to manually correct struck-through bar lines later, or to regroup brackets for "parts".

If you create your composition in Samplitude, set score settings here in order to be able to edit and navigate more easily through the score view. You may then continue editing the score in order to optimize the layout and notation.

Converting to MusicXML format has the advantage (unlike standard MIDI file format) that many manual settings and optimizations of the score (e.g. number of scores, key, voice allocation, pitch, etc.) remain the same.

MPE

Samplitude supports the new MPE (MIDI Polyphonic Expression) expansion. MPE is a new control standard for synths which expands the range of possible expressions which can be produced with these instruments. Unlike OSC, MPE is not a new protocol; rather, it is standard MIDI which switches channels for each individual note.

When MIDI commands for controlling sound parameters (CC, NRPN), or pitch wheel commands for controlling the pitch, are sent to the synthesizer, these are normally applied to all of the notes on a channel. There is no way to assign different pitches or sound changes to individual notes when using multidimensional devices. With MPE, these limitations are removed. This is done by assigning each note to its own channel so that controller and pitch wheel commands can be applied specifically to the note in question.

This enables MIDI controllers which support MPE to produce a much wider range of expressions than a normal keyboard. In addition to aftertouch (the force applied to a key after it is pressed), MPE also offers the ability to "bend" each individual pitch separately to create vibrato and glissandi. An additional Expression Parameter (CC74) is also generated based on the position of the player's finger on the key. Some examples of MPE-compatible MIDI controllers include the Roli SeaBoard (or its mini version, Roli Blocks), the LinnStrument by Roger Linn, and Soundplane by Madrona Labs.

MPE also requires a compatible synthesizer, that is, one capable of generating separate, unique synth voices for each MIDI channel. There are many manufacturers who offer MPE-compatible VST synths. An extensive list is available on the website of keyboard manufacturer Roli Labs.

The key properties that MPE-compatible controllers and synths have are:

- For each Note On event, a new channel is created for the note. The channel is not made available for new notes until a Note Off event is sent for the current note. Since at least one other channel (Master Channel) is needed for global control commands such as switch program and global pitch, the maximum possible polyphony is 15 individual voices.
- CC74, pitch wheel and channel pressure (aftertouch) commands send to the assigned channel are applied to the note in question; CC74 is reserved as a third control dimension for MPE synthesizers (MPE Timbre).
- The pitch bend range that a MIDI note can be pitched up or down using the pitch wheel is set to 48 (+/- 4 octaves). Compared to the normal +/- 2 semitones, this seems like a huge range. But the idea is to give you the greatest possible pitch range to create glissandi from any note.

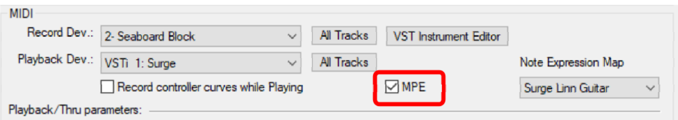
The MPE specification includes other rules, such as those for handling key pressure, for the Master Channel, and for what happens when the polyphony is exceeded.

These rules, however, are not particularly relevant to MPE support in Samplitude. To learn more about them, visit www.midi.org https://www.midi.org/articles-old/midi-polyphonic-expression-mpe.

MPE track mode

In order for Samplitude to be able to interpret incoming MIDI data as MPE, and in order to be able to edit it as such, the track must be set to MPE mode.

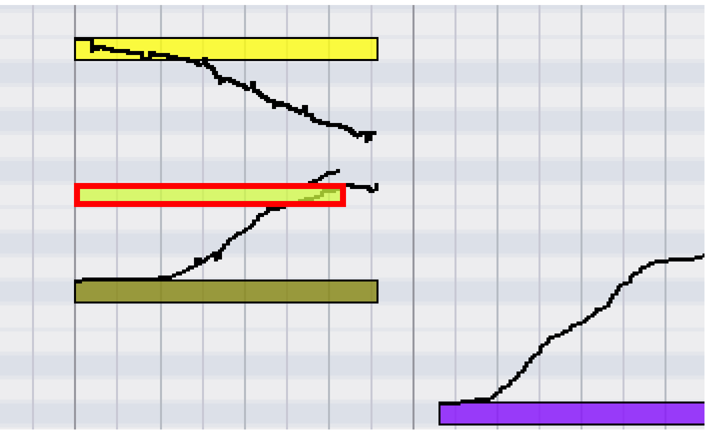
To do this, enable the option **MPE** in the track settings or in the MIDI section of the Track Editor.



The MIDI editor in MPE mode

The MIDI editor's MPE mode has some special features which make it easier to edit MPE MIDI.

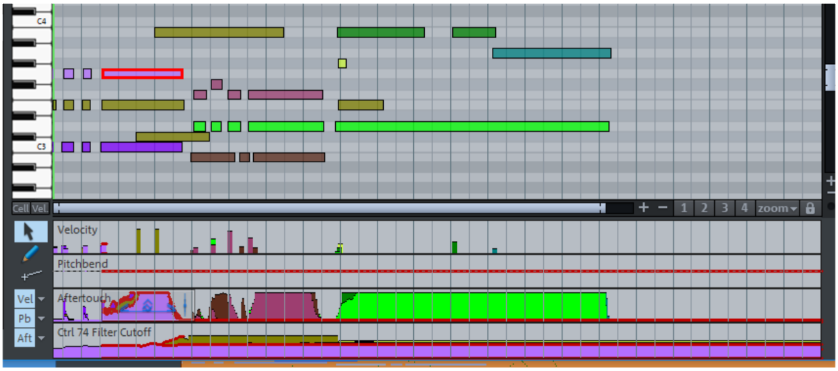
- Thanks to an extremely large pitch bend range comprising 48 semitones, even very small adjustments to the pitch wheel create audible changes to the pitch. Since these tiny changes are not easy to see in the Controller Editor below, the pitch envelope is displayed as an additional curve on the notes.



- When you move a note, the controller/pitch wheel data associated with that note (that is, the data located between a specific Note On/Note Off event on a specific channel) are moved as well.

Tip: To help keep track of which controller data belongs to which note, we recommend enabling the **Use MIDI Channel Coloring** option (located in the Options menu in the MIDI Editor).

- The channel in the foreground of the controller or note is always the one selected last.



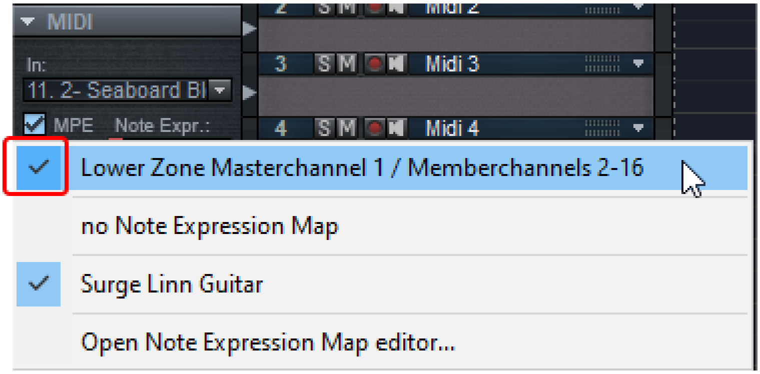
- To prevent this being moved when you drag a selection, hold down the **Ctrl key** when dragging the selection. The selected channel in the foreground is then maintained and only events on this channel are additionally selected. If you want to select events on other channels too, additionally hold down the **Shift key**.
- To make it easier to edit very small pitch bend curves, you can zoom in vertically on the controller curve by holding down Ctrl and using the mouse wheel.

Master Channel

A Master Channel is defined in MPE Standard as well. Controllers which are sent over this channel affect all notes, just like normal MIDI. Switch program commands must be sent over this channel as well.

MIDI Channel 1 is set as the Master Channel (MPE Lower Channel) in Samplitude by default. If you do not need a Master Channel, you can disable it, which will increase the maximum polyphony to 16 voices. You can also disable the Master Channel if you are using Note Expressions (see below).

You can disable the Master Channel in the Note Expression menu to the left of the MPE options in the Track Editor, or under "Note Expression Map" in the Track Options.



VST3 Note Expressions

Note Expressions are another way to control VSTi synthesizers with sound parameters note by note and hence to enable more expressive ways of playing. These are an extension to the VST3 standard. Note Expressions are currently only available for certain synthesizers by Steinberg.

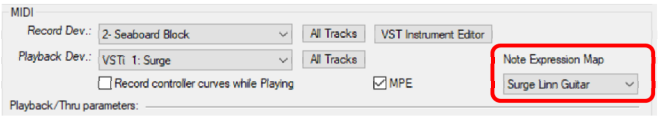
The plug-in provides a specific number of sound parameters which can be assigned as Note Expressions to MIDI controllers via the Note Expression Map in Samplitude, which then apply only to individual notes.

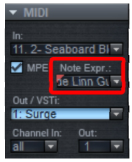
Differences between MPE and VST3 Note Expressions:

- MPE is a MIDI protocol in which notes played at the same time are assigned to different channels when using polyphonic playing. The synthesizer must be capable of receiving notes from different channels and of processing the MIDI controller commands for these notes separately by channel.
- VST3 Note Expressions are an extension to the VST protocol. Samplitude receives MIDI notes on various channels with corresponding controller values and sends the notes to the synthesizer on a channel. The Expression Values assigned to the controllers for each note are also sent over the VST interface to the synthesizer.
- As such, MPE cannot be used to control multitimbral instruments, but Note Expressions can.

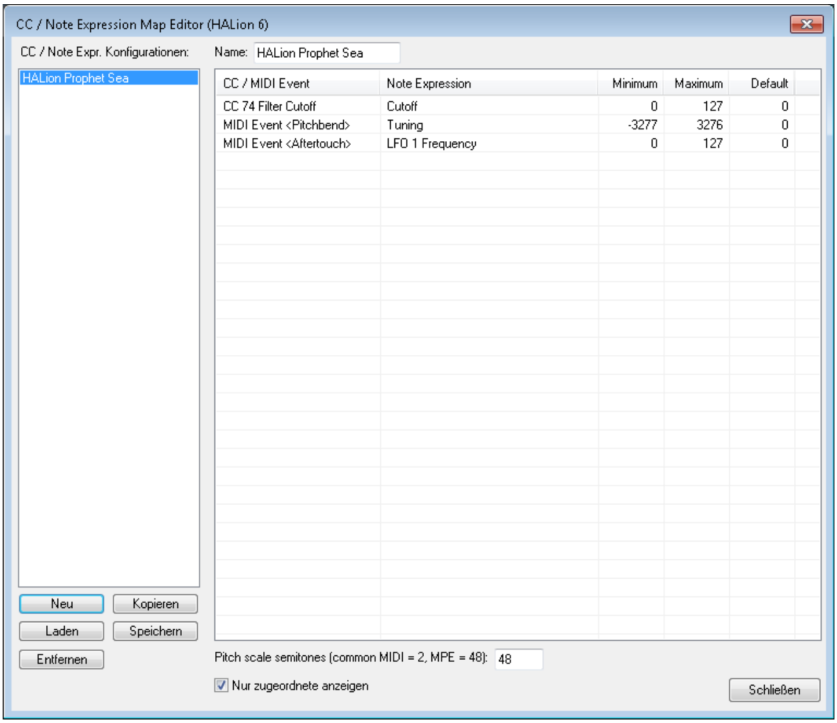
Note Expression Map Editor

You can open the Note Expression Map Editor in the Note Expression menu to the left of the MPE options in the Track Editor, or under "Note Expression Map" in the Track Options.





1. Load a VST3 instrument which supports Note Expressions onto the track. Load a preset which has Note Expressions assigned in it as modulation sources, or make these assignments yourself in the VST instrument.
2. Now, open the Note Expression Map Editor and click "New" to create a new Note Expression Map. This map is now valid for the loaded patch in this specific plug-in.



3. Give the map a descriptive name (e.g. Synth_Preset).
4. Assignments are made on the right. The three standard MPE parameters CC74 (timbre), pitchbend and aftertouch are preset. If you want to assigned the Note Expressions to another controller, disable "Only show assigned".

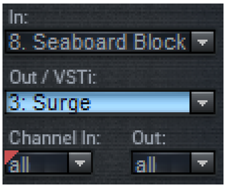
5. Now, click on the corresponding controller in the Note Expression column and select a parameter from the list to assign it to the controller.
6. To create a new map based on an existing one, select a map from the list on the left and click "Copy". "Load" and "Save" let you load a map from a file or save a map to a file, respectively. "Delete" removes the selected map from the list.
7. Under "Pitch scale semitones", you can specify what pitch bend range your keyboard uses. MPE controllers use 48, normal keyboards usually use 2.

Note: The Note Expression parameters depend on the synthesizer's patch settings and are made available by the VSTi when the Note Expression Map Editor is opened (not before). It is therefore important to follow this order: first program the patch settings in the synth, then select the corresponding map!

A note about MPE and Note Expressions

- When using Note Expressions, be sure to switch the MIDI Out channel from "All" to a specific channel (e.g. Channel 1) if you are using a multitimbral synth such as Steinberg HALion which is capable of playing multiple patches at the same time, each of which respond only to specific MIDI channels.

- When using MPE controllers, it is absolutely important to make sure that all channels are selected for use under MIDI In.



- If you do not have an MPE-capable keyboard, you can still use the advanced capabilities of MPE and Note Expressions by manually changing the channels for notes and inserting the corresponding controller values in MPE mode when the MIDI Editor is in listen mode (view page 365).
- If you are recording with a normal keyboard which normally sends notes through Channel 1, be sure to disable the Master Channel or select a different channel on the keyboard.

MIDI Editor Keyboard Shortcuts

Apart from few exceptions such as "Space" for stop/playback, you can set up the keyboard shortcuts any way you like. To do this, open the editor for keyboard shortcuts, menus and mouse through the MIDI menu "Shortcuts > Edit Shortcuts". For some basic functions such as scrolling and zooming, all keyboard shortcuts from the VIP are used. You can explicitly define shortcuts for all commands that are available in the MIDI Editor menu.

Note: The mouse wheel settings are also applied from the arranger, as is the switch for temporary zoom mode (special) which is predefined with "Z".

Play/Stop	Space
Close MIDI Editor (discard changes)	Esc

File

Import MIDI	Ctrl + I
Export MIDI	Ctrl + E

Edit

Undo	Ctrl + Z
Redo	Ctrl + Y
Cut	Ctrl + X
Copy	Ctrl + C
Insert	Ctrl + V
Duplicate	Ctrl + D
Select all	Ctrl + A
Create pattern from selection	Ctrl + Shift + P
Delete selected MIDI events	Backspace, Del
Delete all MIDI data	Ctrl + Backspace, Ctrl + Del
Select previous MIDI event	Arrow left
Select next MIDI event	Arrow right

MIDI Functions

Legato	Ctrl + L
Quantize notes	Ctrl + Q
Quantization settings	Alt + Q
Mute notes	Ctrl + M

Options

Scroll mode	F
Play clicked notes	Alt + P
Quantization grid active	Ctrl + G
Show quantization grid	Alt + G
MIDI Object Editor...	Ctrl + O
Audition Panic-Played Edit Notes	Ctrl + F

Mouse Mode

Selection	1
Draw	2
Draw drums	3
Draw pattern	4
Change velocity	5
Delete	6
Zoom Tool	7
Compile notes	8
Split notes	9
Mute notes (Mute Mode)	M
Select Velocity/Controller	Ctrl + 1
Draw Velocity/Controller	Ctrl + 2
Draw Velocity/Controller as a line	Ctrl + 3

View

Event list	Alt + L
Velocity/Controller Editor	Alt + V

Notes

Accidental #	#

Keyboard Shortcuts

Select previous event (exclusive)	Arrow left
Select next event	Arrow right
Add previous event to selection	Shift + Arrow left
Add next event to selection	Shift + Arrow right
Event pitch up	Arrow up
Event pitch down	Arrow down
Snap event to left	Ctrl + Alt + 1
Snap event to right	Ctrl + Alt + 2
Select next grid quantization value	Alt + Arrow down
Select previous grid quantization value	Alt + Arrow up
Select next length quantization value	Alt + Arrow right
Select previous length quantization value	Alt + Arrow left
Move playback marker forward	Page Down
Move playback marker backward	Page Up
Playback marker to next bar	Ctrl + Page Down
Playback marker to previous bar	Ctrl + Page Up

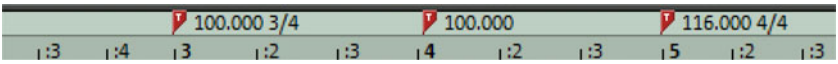
Tempo editing

Tempo and beat change

Tempo changes can be defined by means of the tempo marker in the marker bar or the

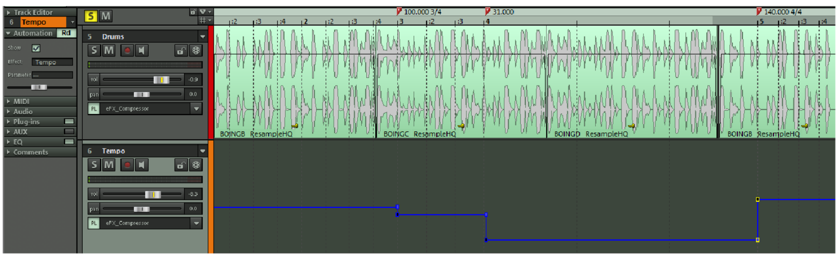
marker button. Tempo changes are displayed in the marker bar.

The Tempo Map (view page 399) can be opened through "Edit" > „Tempo" > „Tempo Map". The tempo map displays all tempo markers and thereby determines the musical position for each time position. The tempo map defines the bar grid.



This means that the bar grid is made up of all the tempo markers. Samplitude offers you convenient options for adjusting the bar grid to existing material (audio/video/MIDI).

You can also display and edit a graphic display in the tempo track (view page 403). A tempo track contains the project tempo map as an automation curve. Each automation point of the curve corresponds to a tempo marker.



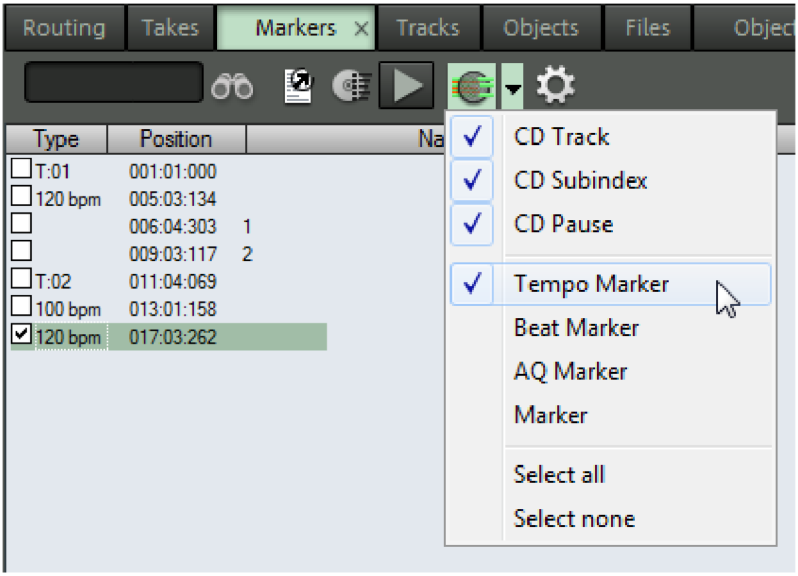
BPM marker, tempo curve (>) and beat change (3/8) - under the tempo track.

In addition to the display in the tempo map dialog and the tempo track, the current values for tempo (BPM) and the beat type at the playback marker position are displayed in the transport console (view page 90).

Tempo Markers

Tempo markers signify a tempo change at a specific position in the project. They appear either as BPM markers or as Grid position markers according to the Tempo Map mode (view page 401).

You can also change the time signature when creating BPM markers and grid markers. Set the Tempo marker directly using the menu or indirectly by manipulating the tempo in the tempo track with the mouse. Another way of accessing and editing the Tempo marker is by using the Marker Manager (view page 184).



BPM Markers (red)

You can define tempo changes for each musical position directly in the project window using BPM markers. When setting a BPM marker, a new tempo is defined at the corresponding position. A new project first has a single "Master" tempo which can be defined in the transport console or in the project settings (keyboard shortcut: "I").

This master tempo affects either the entire project or the start of the project up to the first BPM marker.

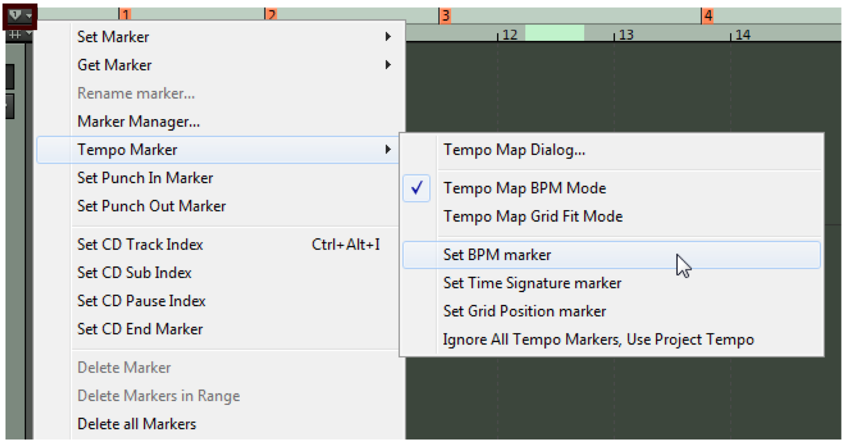
Exception: If the tempo should be interpolated to the first tempo marker, the master tempo is only effective directly at the project start in order to go linearly to the tempo of the next BPM marker.

You can define tempo changes at any time and for each musical position (even between beats) directly in the project window using BPM markers or the corresponding tempo curve points:

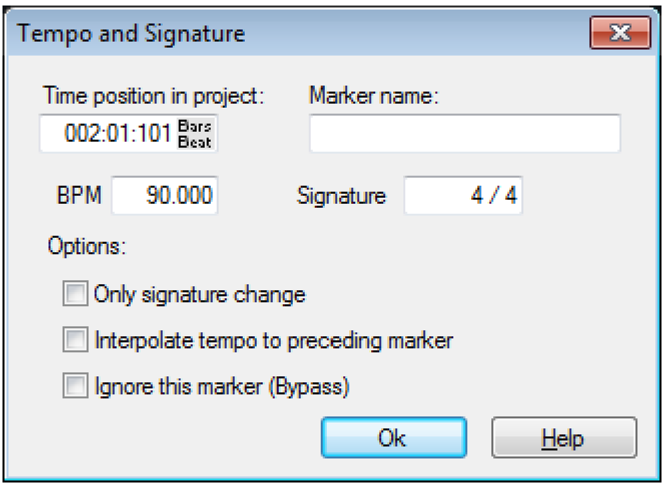
Step 1: Place the playback marker at the position where the tempo change should take place.

Step 2: Right-click on the marker bar to open the marker menu.

Step 3: Go to "Tempo markers" > "Set tempo change"



Step 4: Enter the desired tempo change in BPM (beats per minute) in the "Tempo and Beat Type" dialog and confirm by clicking "OK".



The tempo markers are set to the next snap point to the playback position when the grid is active. By dragging the tempo markers and holding down the "Alt" key, you can temporarily deactivate the grid function.

Note: In order to use tempo markers to adjust audio objects in your arrangement to match tempo changes, musical tempo adjustment (view page 408) needs to be activated for the object.

Time Signature Markers (blue)

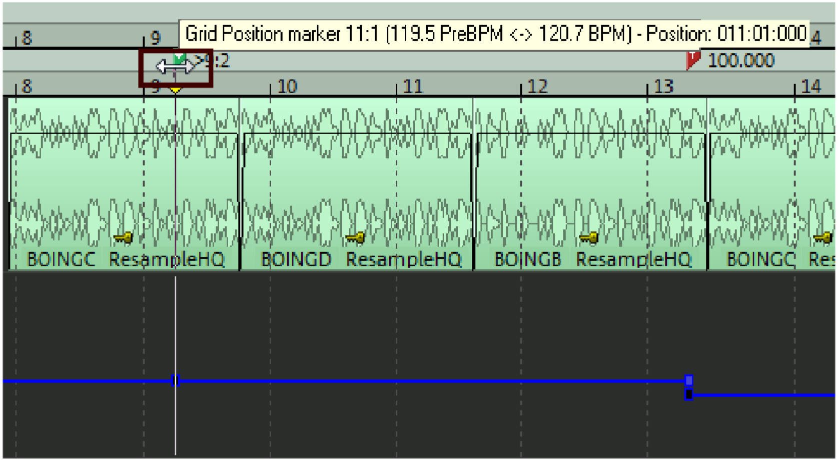
Bar markers change the time signature from the marker position, e. g. 4/4 beat to 3/4.

This means you can only insert bar markers at the borders of bars. They must also be positioned at bar borders when moved.

Grid Position Markers (green)

Grid position markers assign a specific musical position to a specific time position.

If you move a grid position marker, you'll see how the beat grid is immediately adjusted. In the tempo track (view page 403) you can apply the effects of the beat grid changes to the tempo.

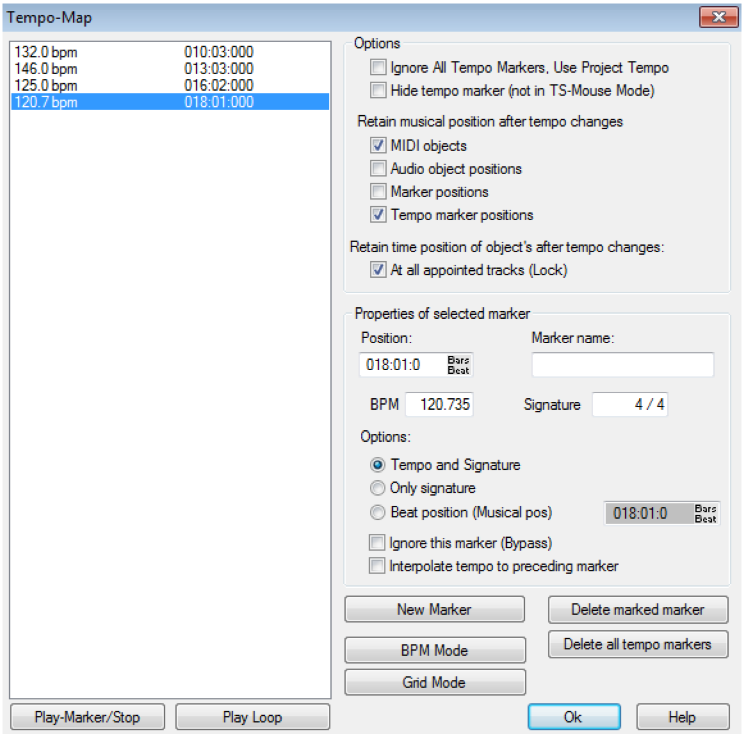


A grid position marker defines the tempo indirectly, causing the tempo before the marker to be adjusted in such a way as to reach the desired musical position exactly at the marker position. This way, the bar frame/grid and MIDI events can be easily synchronized with existing audio material.

Tempo Map Dialog

Step 1: Set the playback marker to the position at which the tempo or beat changes should occur.

Step 2: Open the tempo map dialog with the command in the marker menu (right-click on the marker bar) or through "Edit" > "Tempo".



Step 3: Now click on "New Marker". A marker is inserted at the current position.

You can now use the marker options to set the properties of this marker. Now make the new entries, e. g. the tempo in BPM for a tempo marker (view page 396) or the new time signature for a bar marker (view page 398).

Note: Be aware that the settings are only valid for the markers currently selected in the marker list. For example, you cannot set marker properties such as type or tempo after a new marker has been created.

A bar marker is always placed at the start of a new bar. If the playback marker is at a different position, the marker position is automatically moved to the beginning of the next bar.

Ignore all tempo markers, use project tempo

This option allows you to ignore any previously created tempo markers in your project so that only the project tempo is observed.

This occurs automatically for grid position markers if illogical bar positions are created while moving, e. g. if grid position markers are exchanged (bar 20 before bar 19) or are moved in such a way that this bar position originating from a previous tempo marker cannot be met by tempo interpolation.

Hide Tempo Marker: This option ensures that all tempo markers in the arranger are no longer shown. If you are in the pitch shift/timestretch mouse mode, this option is ignored.

Retain musical position after tempo changes

If a tempo marker is changed or a grid position marker (view page 398) is moved, this influences the following markers and audio/MIDI objects in the virtual project.

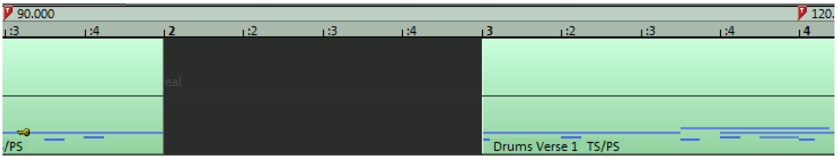
There are two main possibilities: The time position either remains constant, or the musical position remains.

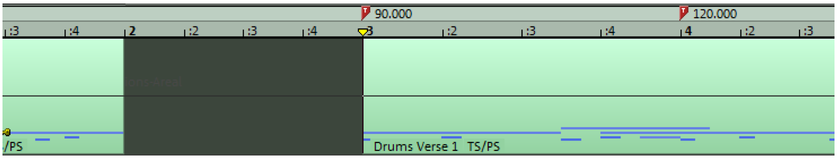
The time position in the virtual project is the absolute position, the musical position is flexible and is displayed by the grid.

Maintaining the musical grid position means that objects or markers are adapted to the newly created grid and their absolute time position in the project changes. If the time position is maintained, the musical position changes accordingly.

When changing the tempo markers, the behavior of markers, audio and MIDI objects is controlled separately. By default, the musical grid position of MIDI objects and tempo markers is retained and the time position of markers and audio objects remains constant.

If you drag a tempo marker while holding down the Alt key in Grid Adjustment Mode, the time position of the object remains unchanged.





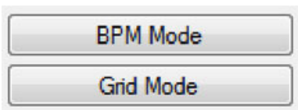
In this example, the first tempo marker in the second image has been changed. The MIDI object and the second tempo marker were moved correspondingly so that the musical position (3:01:000) was maintained.

If you generally want to keep the time position of objects during a tempo change, select the option „On all locked tracks“ and activate the button in the track header  to lock all the objects on the track.

Tempo map modes BPM mode/Snap mode

In the Tempo Map dialog (view page 399) you can choose between two modes: BPM mode and Snap mode.

- In BPM mode you work exclusively with BPM markers and Time signature changes.
- In Snap mode you work exclusively with grid position markers and time signature changes.



You can toggle between BPM markers and grid position markers at any time. The tempo markers remain the same, but change their type. The effective tempo map and beat grid remain unchanged. For example, first adjust the bar grid in grid mode to a MIDI recording with many tempo changes. Now switch on the BPM mode and insert musical tempo changes for single sections.

Note about working in BPM mode: By holding down the "Alt" key in the time stretch mouse mode, you can also move a temporary grid position marker while dragging with the mouse in the timeline.

Grid Tapping

You can set grid position markers in the tempo map grid fit mode during playback.

Step 1: Assign the menu command „Insert grid position marker“ (view page 654) to a keyboard command of your choice, e. g. „+“.

Step 2: Now press „+“ in rhythm with the music during playback to "tap in" the beat grid and tempo map.

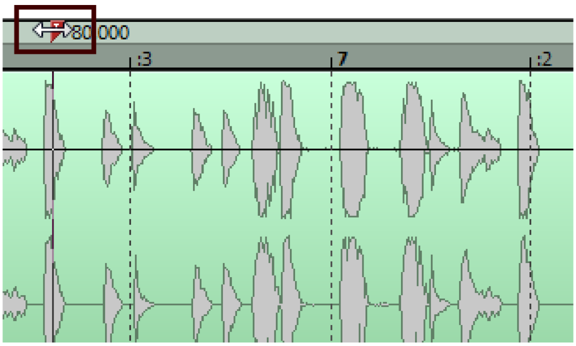
The following will happen: At the time of pressing the key, the nearest bar position of the selected bar grid will move to the current playback position and it will be marked with a grid position marker.

> **Tip:** To detect the tempo, work in the bar grid (entire bar borders) first. In a second step, you can edit slight tempo fluctuations in the "beat" grid.

Manipulating the Tempo Marker in Timestretch Mouse Mode

The timestretch mouse mode simplifies working with tempo markers. Select "Pitchshift/Timestretch" mode from the toolbar. Create BPM markers by directly clicking the desired playback position and holding down "Shift". You can adapt this tempo by moving the mouse vertically while holding down the Shift key.

You can create grid position markers in "Timestretch" mouse mode by clicking the desired playback position and holding down "Alt". You can move them using "Alt" + horizontal mouse movement, e. g. to manipulate the bar grid and adapt it to existing audio events.



Tips - Examples of Use

- Tempo markers can be placed when composing in the MIDI Editor. Subsequent MIDI Objects and tempo markers retain their musical position (advanced setting).
- The grid can be adjusted to available audio events, e.g. assigning beat numbers to certain time positions. You can either use the "Set new grid position marker" menu command to create a marker at a position within the project and assign the corresponding bar position, or use "Alt" + mouse click on the respective beat grid position and, while holding down the mouse button, immediately move it to the required time position (e.g. to the beginning of an object).
- If the project includes MIDI data, this will be automatically adjusted (preset). The newly created tempo grid is used in the MIDI Editor for subsequent editing.
- When working with MIDI files and complex tempo changes, you can switch off the tempo map before starting to record the new MIDI data ("Ignore all tempo markers, use project tempo"). After the recording has finished you will be able to switch on the tempo map once again, which makes the newly recorded MIDI data adjust itself automatically.

Tempo Track

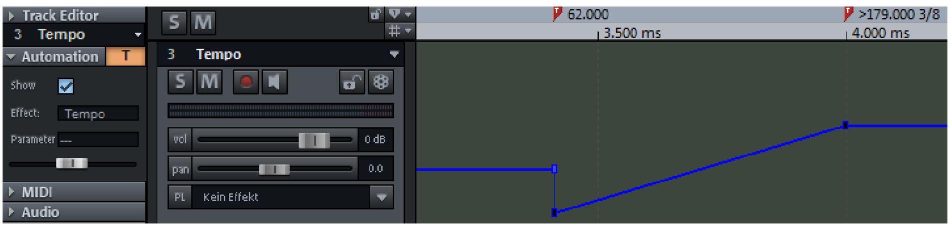
The Tempo Track can be used to display an overview of the tempo and time changes in your project in a track of its own. The Tempo Track is a graphic display of the project's tempo map (view page 399) as an automation curve (view page 428). Each automation point of the curve corresponds to a tempo marker.

The Tempo Track header provides quick access to the most important commands for placing and editing tempo markers.

Note: *You still have the option of displaying the "tempo" automation curve as one of many automation curves in any other track, instead of displaying it in a separate Tempo Track. When you create a Tempo Track, the tempo automation curve in the other track is removed.*

Creating Tempo Tracks

To create a Tempo Track, select "Track" > "Insert new Tracks" > "New Tempo Track".



Changes to tempo markers are automatically transferred to the Tempo Track and vice versa. Keep in mind that the tempo track can only be edited in BPM mode; in snap mode it serves solely as a display.

The tempo automation behaves differently than track-based automation curves. Automation points are coupled with evenly increasing/decreasing tempo changes and the pencil of the automation drawing mode doesn't create any new automation points as you drag, only when you click.

Note: In the tempo track the beat grid is constantly active (regardless of the system option "Program > General > Use snap for automation curve points"). As usual the snap can be temporarily disabled using the "Alt" key.

Tempo Track header

1. **Tempo automation on/off:** You can disable tempo automation entirely with this button. This corresponds to the option "Ignore all tempo markers, use project tempo" in the Tempo Map dialog (view page 399).
2. **Current tempo:** The dropdown menu under the arrow corresponds to the one in the transport console and contains a selection of preset tempos, as well as the tap tempo feature.
3. **Tempo Map dialog**
4. Insert new **tempo marker**
5. Insert **signature change**
6. Insert **grid position marker**
7. Switch between Grid mode/BPM mode (view page 401)

Edit in the Tempo track (only BPM mode)

Insert tempo changes

- Double-click on the tempo curve in Universal Mouse Mode:
- In Curve Mode: Click (and drag) on tempo curve.
- In Automation Draw Mode: Click

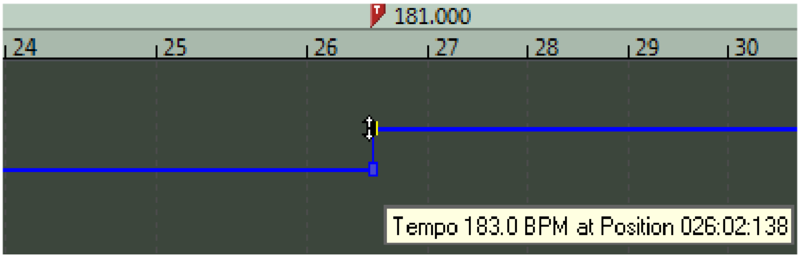
Change existing curve points

- BPM. value and positions can be changed by clicking and dragging curve points. For tempo levels there are always two curve points for one tempo marker, one for position and and one for tempo.
- Alt key: With this modification key you can temporarily deactivate the grid and convert tempo jumps into tempo curves.
- Shift key: With this modification key you can change BPM values and make fine adjustments to them (horizontal lock).
- Ctrl + Shift + Click: This key combination can be used to switch between tempo curve and tempo step.
- Right-click: Right-clicking on a tempo marker opens the „Tempo and Time Signature" dialog (view page 407) where numerical values can be changed.

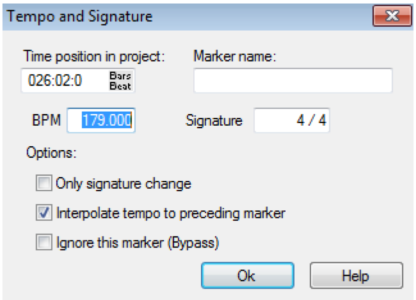
Inserting tempo changes and tempo curves into the tempo track:

Step 1: Double-click (in Universal Mode) or simply click (in Curve Mode or Automation Draw Mode) on the desired position in the tempo curve. In addition to the new curve point a new tempo maker will appear in the marker bar.

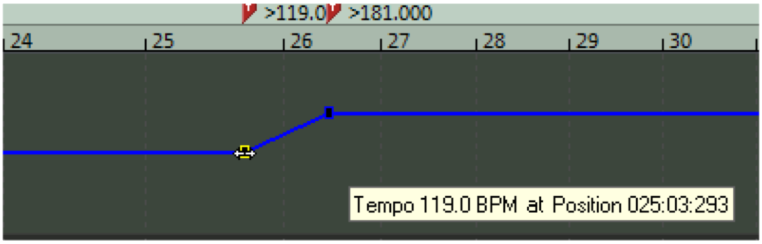
Step 2: Starting at the new curve point, drag the mouse pointer upwards or downwards to change the tempo.



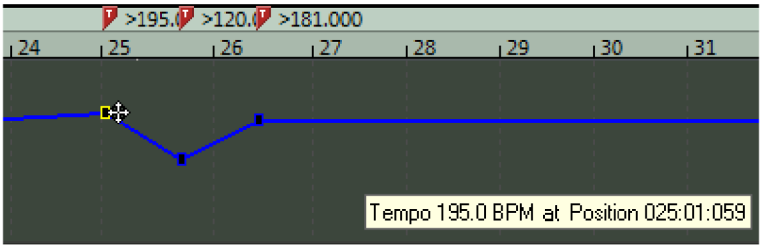
Step 3: Right-click to the open the "Tempo and Time Signature" dialog where you can change the values numerically.



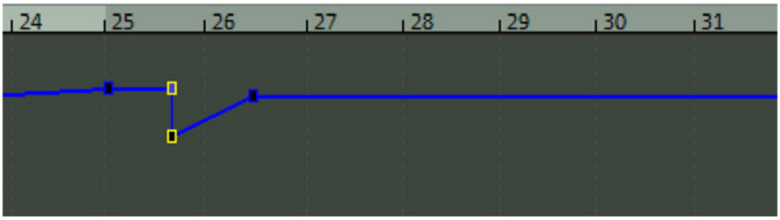
Step 4: If you hold down the Alt key while clicking on a tempo marker, the grid is deactivated and the tempo jumps to the neighboring tempo markers are converted into tempo curves.



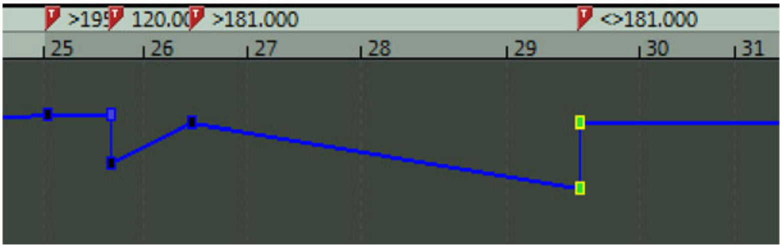
Step 5: Hold down the Ctrl key while double-clicking (in Universal Mode) or simply clicking (in Automation Draw Mode). Now drag on the created BPM marker to set a tempo curve to the previous BPM marker.



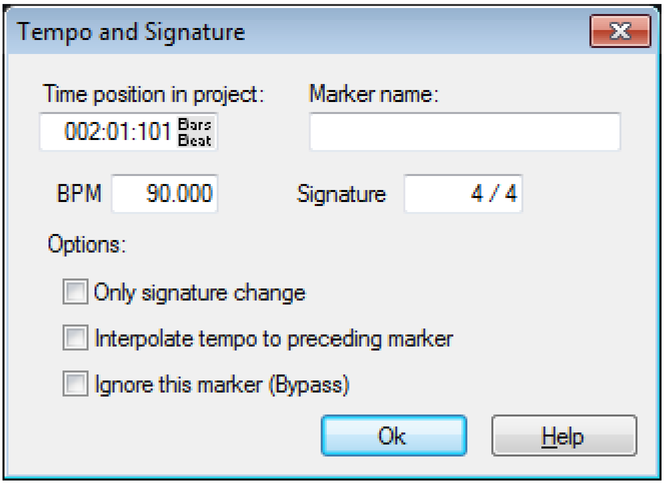
Step 6: To switch back and forth between the tempo curve and the tempo step to the previous tempo marker, click on a tempo marker while holding down the Ctrl and Shift keys.



Step 7: This creates a Ritardando or Accelerando. While holding down the Ctrl key, create a new tempo marker by double-clicking (in Universal Mode) or simply clicking (in Curve Mode or Automation Draw Mode) on the tempo curve. Hold down the mouse button and drag upwards on the created tempo marker for Accelerando, or downwards for Ritardando. This will generate a tempo curve to the previous tempo marker and the corresponding BPM marker will be displayed as a double arrow in the marker bar. The previous tempo is applied again from the newly created tempo marker position.



Tempo and time signature dialog



This dialog appears by double-clicking on an available tempo marker or curve point or if you set a new tempo or bar marker via menu command ("Edit" > "Tempo" > "Insert tempo/beat change").

As in the tempo map dialog, you can also determine the time position in the project as well as the marker name here, enter the desired tempo in BPM (not for grid position marker) and determine or change the time signature. Also you can choose here whether the tempo should be interpolated to the previous tempo marker (Ritardando or Accelerando) and whether this marker should be ignored.

Change Global Tempo

Using the command „Edit“ > „Tempo“ > „Insert Tempo Change...“ You can scale all of the BPM values on selected tempo curve points by a set factor simultaneously.

Musical Tempo Adjustment

With musical tempo adjustment, audio objects are adjusted to match any changes made to the tempo of the project. This means that when you change the tempo of the project, the tempo and time positions of the objects are adjusted accordingly using tempo markers or a tempo curve. The bar position remains untouched while the start positions of objects are moved, and time stretching is applied.

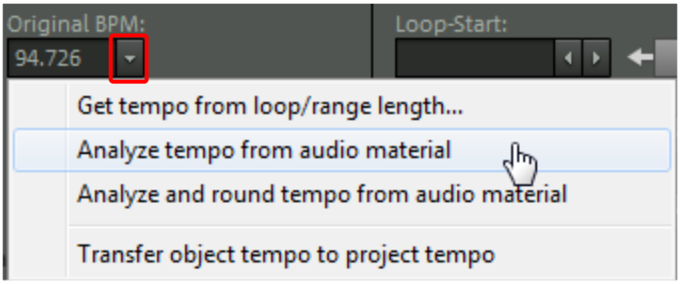
You can activate musical tempo adjustment in an object by placing a checkmark next to it in the Time/Pitch section of the object editor.

Note: Musical tempo adjustment is automatically activated when you load Soundpool samples into a project. If you activate the **BMP sync** preview option (view page 176) when you load audio files via the file manager, tempo adjustment will also be activated for the object.

The **Original BPM** setting used as a basis for the adjustment must be a valid value.

A question mark next to the entry field indicates that the displayed value is a default value and may not be correct.

You can either enter the original tempo manually or let it be calculated. To do this, click on the arrow next to the value.



Get tempo from loop/range length: A range section above the object is used to calculate tempo. Select the option and enter how many quarter notes should be included in the selection.

Analyze tempo from audio material: Melodyne will calculate tempo. Melodyne must be installed for this. You can learn more in Melodyne integration (view page 292).

Musical tempo adjustment cannot be used at the same time as Elastic Audio (view page 810) (pitch automation), ARA (Melodyne integration) (view page 292) or object resampling.

The indicator field in the object will show you which of these options is currently being used:

Elastic Audio (pitch automation).

Musical tempo adjustment: The length and position of the object are changed to match the tempo map of the project. The original BPM value will be used as a reference tempo for the object.

Sample rate adjustment: Adjusts the sample rate of audio objects by using object resampling if the sample rate of the Wave file is different from that of the project. If sample rate adjustment has been carried out automatically, you will be asked if you want to remove it when you activate tempo automation.

Object tempo and pitch are controlled by the Melodyne plug-in (view page 292).

Note

- The positions of object automations and audio quantization markers (view page 694) will be taken into account and changed accordingly.
- In the current program mode, non-relevant control panels (such as stretch length, stretch factor and BMP) are grayed out.
- Remix Agent (view page 710) cannot be run.
- Tempo automation does not work with looped objects or objects played in reverse (view page 153).

VST and ReWire

Samplitude provides the option of integrating software instruments into a VIP according to VSTi standards or through ReWire and then controlling them with MIDI.

VST is an interface for additional audio software that enables VST instruments and effects to be used as plug-ins in Samplitude. VST instruments create sounds virtually and can imitate real instruments and even synthesize entirely new sounds. They are controlled by MIDI and can be played using a MIDI compatible keyboard.

VST instruments can be accessed as audio track inputs and as mixer channels. In this way, the audio signals of a VST instrument can be further edited with all the options offered by the mixer, including the EQ, effects and routing. The maximum number of plug-ins is only limited by the performance of the system processor. You can increase the number of applied VSTs almost limitlessly by using the integrated freeze function

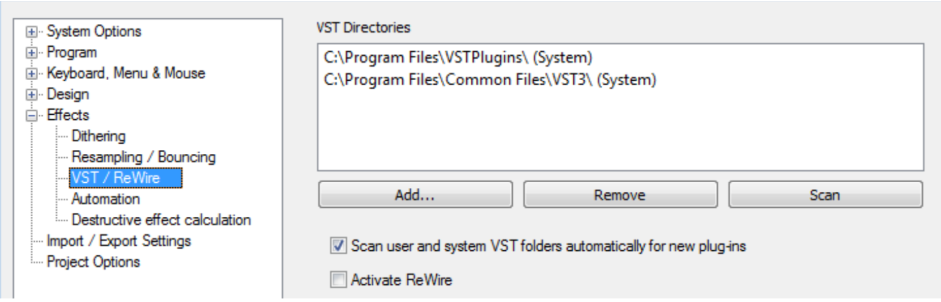
ReWire is a software protocol that allows you to remotely control another program from within Samplitude. For example, you can virtually connect a different DAW to Samplitude and exchange data between the programs directly. This does away with the need for importing audio and MIDI files.

ReWire is generally used for sample-exact, real-time transfer of audio channels between two programs. Both programs can be connected through the same sound card. The transport functions of the applications (such as playback and forward/reverse) are connected with ReWire. If ASIO drivers are being used, sounds from ReWire-compatible applications can be assigned to different sound card outputs.

Installation of VST Plug-Ins

The VST plug-ins provided by Samplitude are detected automatically and can be used immediately. To use third-party plug-ins in Samplitude you must tell the program in which folder they have been installed. To do this, proceed as follows:

1. Install each VST plug-in according to the instructions provided by the manufacturer. In most cases the plug-in installer suggests a default folder (e.g. C:\Program Files\VST): these System VST folders are included in the plug-in scan by default. You can also use any other folder. If you already have VST plug-ins installed on your system, it is best to use the existing folder.
2. Open the system settings (Key Y) and go to "Effects > VST/Rewire". Under the VST plug-in folder list, click "Add" and enter the folder path.



3. To start the plug-in scan, click "Scan" and choose "Scan selected VST folders" from the menu. This scan may take some time for many installed plug-ins and instruments. Not only are all the plug-ins imported, but they are also checked for usability within the program. Incompatible or incorrectly installed plug-ins and those that cause the plug-in search to crash are also included in the list and flagged as unusable, so that they are skipped during the next scan and cannot cause problems again. To scan these plug-ins again, use the menu item "Test failed plug-ins again".
4. The newly found plug-ins can be used immediately.

Hint For a complete reset of the VST settings and a new scan of all plug-ins, delete the file "VSTPlugins.ini" in the folder C:\ProgramData\Samplitude. All paths will have to be newly entered.

If you activate the option "Scan user and system VST folders automatically for new plug-ins", the list of plug-ins is automatically updated after every program startup when the track settings or plug-in browser are opened for the first time. This means that the program searches for newly installed plug-ins and removes uninstalled plug-ins from the list. This search is performed only once per working session, and the next time you access the track settings it'll open immediately.

You can also specify and scan any number of plug-in folders.

Hint: If you add VST plug-in subfolders into the main VST folder (e.g. "Equalizer", "Filters", "Modulation", etc.), these will also be displayed as subfolders in the plug-in browser.

Hint: The command for a new plug-in scan is also in the menu that you can open with an arrow button on each plug-in slot.

Load software instruments

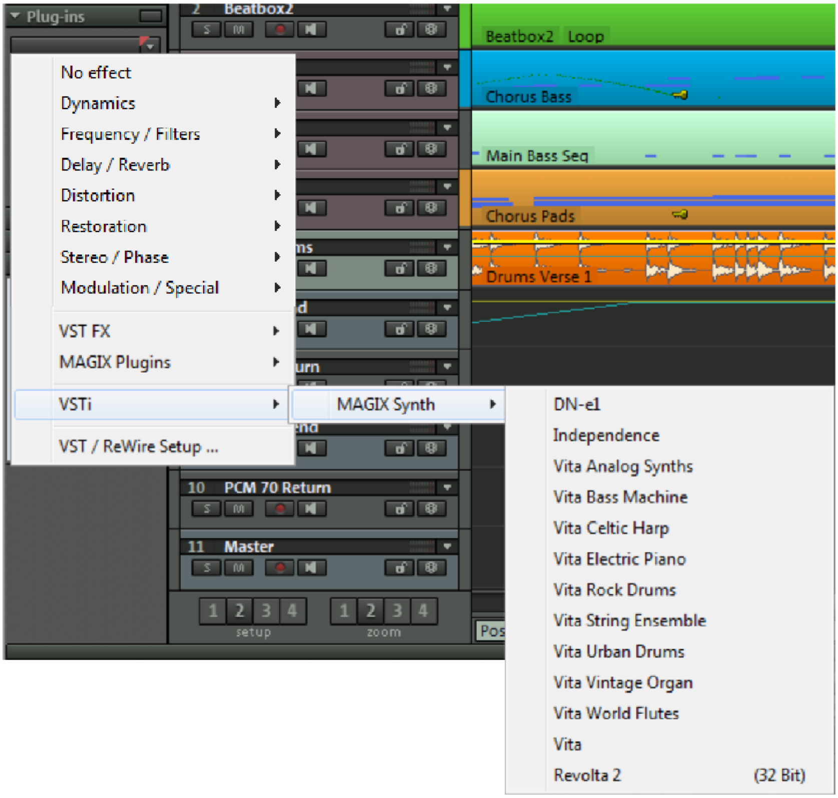
You can also assign a software instrument to each track; the instrument is selected as a MIDI playback device. Selected instruments or their individual outputs are shown directly in the first VSTi plug-in/insert slot of the track header and track editor and can also be muted (left click) and opened (right click) from there.

Load an instrument by selecting a "New instrument" in the project window as a track output. This can happen at various positions:

- In the MIDI section of the track editor: "Out" slot -> new instrument

This is the standard method for loading and routing a software instrument.

- In the playback device context menu, (right click "Mute -> MIDI -> New instrument)
- Plug-ins slot ->Plugin Browser in the track editor



Software instruments may also be created in the mixer:

- Insert effect slot of the mixer ->Plugin Browser

This option is located in the top insert slot of the individual mixer channels.

Alternatively, you can also install and manage software instruments in the VSTi rack. To do so, open the VSTi manager (view page 195).

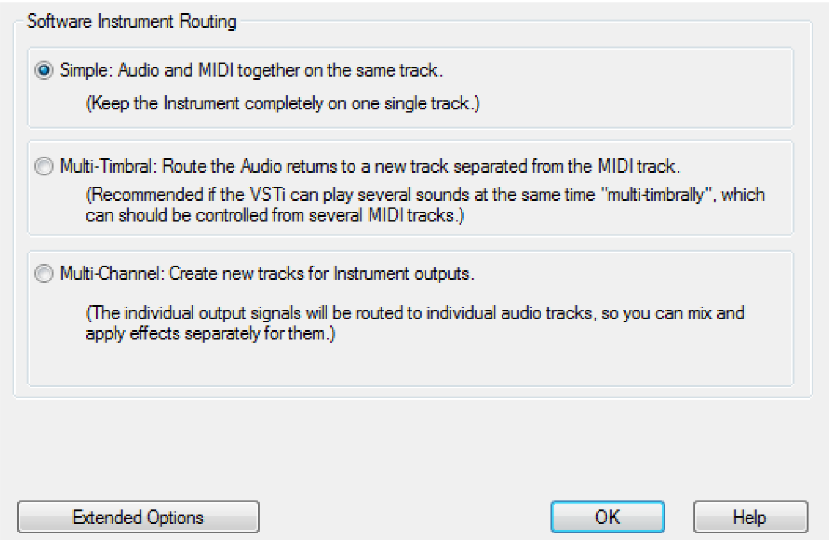
Note: You can load plug-ins In the master section of the mixer, just like in the mixer tracks. Click on the arrow to the right beside one of the plug-in slots and select the desired plug-in from the menu.

Loading routing settings with software instruments

An instrument may also be assigned to each track as a MIDI output device..

Audio output channels may also be routed from software instruments to any VIP track. MIDI (send) and audio (return) signals from a software instrument may be present together on one track (but this is not absolutely necessary). Samplitude is fully flexible in terms of individual routing configurations.

When installing a software instrument, the "Routing for multi-channel instruments" dialog opens. This is also available via the VSTi manager under "Routing settings".



Note: Press "Extended options" to specify that this dialog is always displayed when inserting a new software instrument into a track or when adding a track for individual output in the VSTi manager.

Simple mono or stereo software instruments may be installed completely in the current track. In this case, the track sends MIDI to the instrument and receives the audio signal from the instrument. Select the "Simple: Audio and MIDI together on the same track" configuration. Now all audio signals will be routed to the track that contains the instrument. Multiple outputs will be mixed together before the mixer effects. You may specify that only the first stereo output is routed to the current track via the advanced options. You may specify that only the first stereo output is routed to the current track via the advanced options.

Multi-timbral: Route the audio to a new track separated from the MIDI track

Several MIDI tracks are typically used for so-called "multi-timbral synthesizers", which may play several sounds on different MIDI channels simultaneously, in which case each device controls a specific sound program (part) on a specified MIDI channel. You can specify in the extended options that only the first stereo output is routed to the current track. The audio return track may also be hidden in the arranger. This setting is useful if individual outputs of an instrument are controlled by a single MIDI file, which therefore do not contain objects or information in the arranger window.

Multi-channel: Create new tracks for instrument outputs

Choose this option to automatically create new tracks for all VSTi audio outputs. The newly created tracks are named accordingly. The mono/stereo configuration is assumed from Samplitude (default).

Note: Individual outputs may also be routed to separate tracks. To display hidden tracks in the arranger window again, use the track manager.

Advanced options:

Stereo/mono (standard): Information delivered from the plug-in is used for routing.

All as mono: This option forces individual outputs to be treated as mono outputs.

All as stereo: This option forces individual outputs to be treated as stereo outputs.

Audio/MIDI combined (tracks also send MIDI): Activate this check box to automatically route the MIDI output for each individual track to this instrument.

Don't show the audio return tracks in the arranger: All newly created output tracks for this instrument are hidden in the arranger but still appear in the mixer window. This setting should be used if an instrument's individual outputs are controlled by a single MIDI file, and therefore do not contain objects or information in the arranger window.

Note: A virtual synthesizer's audio output may normally be created, edited, and mixed in the same track as the MIDI data the instrument is receiving. In some cases, this results in duplicate use of the volume fader which, on the one hand, controls velocity or MIDI volume (CC7), and on the other, this controls the audio level. These are not identical parameters. For instance, you may include a MIDI instrument played with high velocity quietly in the mix and vice versa. Therefore, the volume fader may be assigned differently. To do this, right click the track's volume fader.

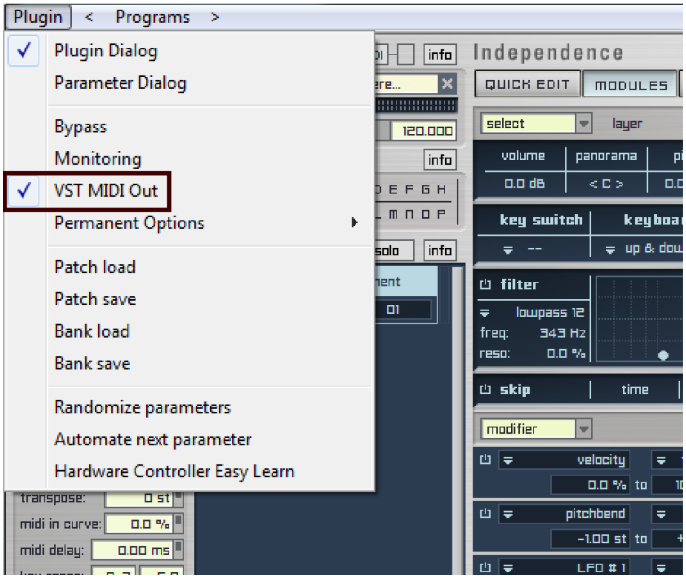
VST MIDI out + Audio out recording

This function has been revised in Samplitude. For the latest information, see the PDF document **Samplitude Pro X7 New Features** in the program folder.

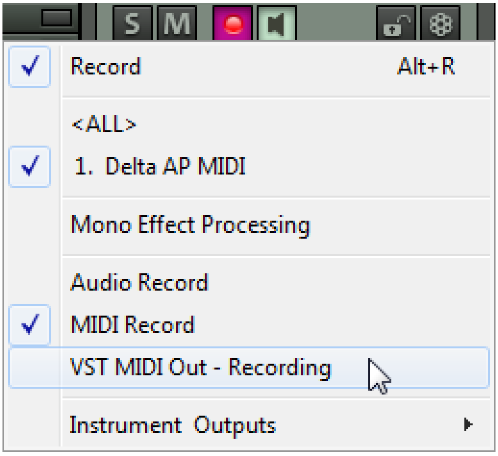
(for "Mixer FX monitoring/Hybrid Engine" only)

You may record VSTi audio outputs for VSTi return tracks by activating recording in these tracks.

"MIDI out" may be activated for every track VST plug-in or VSTi in the plug-in panel menu. This sends and records MIDI data from the VSTi in another track.



By activating the option "VST MIDI out recording" for a track (right click the record button), you can record all received VSTi MIDI out data in the corresponding track. Select the instrument output of the VSTi to be received.



If MIDI thru (view page 52) is activated, this signal may also be routed to another VSTi/VST-FX or external MIDI device. By right clicking the "Moni." button in the transport control, activate the "Automatic MIDI monitoring (thru) at recording start" function.

After recording, apply the track freeze function to visualize the waveform of the recorded track. If you remove the check mark again in front of the VSTi output with a right click on the record button in the track box, the routing connection to the sending VST instrument will be removed.

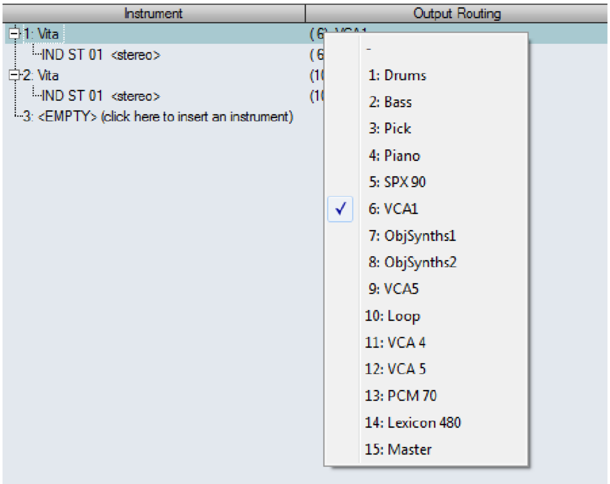
Routing VST instruments using the VSTi manager

There are many flexible methods for signal routing of instruments.

As described in the previous paragraph, an instrument may be fed from several MIDI tracks, and the outputs of an instrument may also be sent to several tracks.

Routing instrument outputs

Instrument outputs can be easily routed by using the VSTi manager. Simply open the VST instrument to view the single outputs and then right-click the "Output routing" column.



Now switch to the track that is to be fed with the respective single output. The assigned output then appears in the target track's plug-in slot.

Alternatively, you can also route directly to the track header of the arranger: To play the output of an already loaded instrument on a specific track, click with the right mouse button on the record button of the track, go to "Instrument outputs" and select the output of the software instrument available in the arranger you want to sound on this track by placing a check mark. Several outputs of an instrument can be routed simultaneously to a track. You can even combine the outputs of different instruments onto one track.

Deactivation of the instrument outputs works in the same way. Remove the check in front of the corresponding output in the "Instrument outputs" submenu.

Note that the routing options of the mixing console may also be used. A track containing the audio return of any instrument may be routed to any submix bus, AUX bus or master.

Routing instrument MIDI inputs

Most software instruments require MIDI data in order to generate an audio output. To route the MIDI output of a track to the input of a VSTi, click on the MIDI button in the Track Editor of the corresponding track and select the VST instrument in the "Out" slot that you want to activate. Already loaded VST instruments are listed in this menu.

To open a completely new instance of an instrument that works independent of any existing instruments within the project, select its name from the "New instrument" submenu. You can differentiate several instances of the same VST plug-ins by the number (index) in front of the name of the software instrument.

Solo playback of VSTi instrument outputs

To preview solo instrument outputs, switch the track that is receiving the signal from the instrument to solo. Even if the MIDI objects, which in effect are responsible for the sound, are not present in the track, you will still hear the selected track solo. Samplitude automatically recognizes the MIDI track that is sending the signal to the output and allows the instrument continue to receive MIDI data from the assigned tracks.

Instruments with multi-channel outputs

The number of available individual outputs generally depends on the instrument, and this may be adjusted via the instrument's settings. Outputs may be available in stereo or mono.

Here are two examples of typical handling of individual outputs:

Addressing an instrument on a MIDI channel and distributing the sounds across several tracks

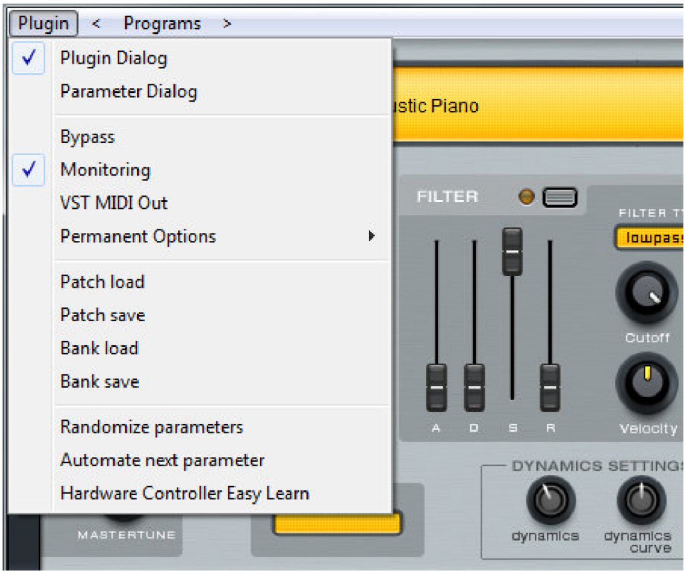
This approach is ideal for drum samplers to address an entire drum set and simultaneously mix and edit the drum sounds individually.

Addressing an instrument on several MIDI channels

Several MIDI tracks are typically used for so-called "multi-timbral synthesizers", which may play several sounds on different MIDI channels simultaneously, in which case each device controls a specific sound program (part) on a specified MIDI channel. The advantage of this is that these instruments only require one instance of the plug-in for several sounds. Multi-timbral software instruments often have separate audio outputs. Don't forget to route the individual parts to the desired outputs within the VST instrument (panel).

Plug-in panel – graphical interface

The plug-in dialog (panel), the graphic interface, opens when an instrument is loaded and can be opened again later by right clicking on the plug-in slot which displays the instrument's name. For instrument plug-ins, you can also open the panel via "Track -> VST instrument editor".



The graphical interface opens by default. The ReWire client application will open when ReWire instruments are selected.

"Programs" features the presets for the corresponding plug-in.

Besides "Programs", you will also find "Sidechain input (view page 426)" for those plug-ins that feature the "Sidechain" function.

Plug-in Menu Functions

Bypass: Deactivates the instrument and mutes it. It is important to note that some instruments require processor power when they are bypassed. For this reason you should remove these instruments completely if you don't need them anymore.

Monitoring: This option must be switched on if you want to hear the instrument when it is being played live or recorded. If you want to have have monitoring active when the track's REC button is on, select the "Tape Monitoring" option under "System Options" > "Audio System".

VST MIDI Out - Recording: In the Plug-in Panel menu you can activate "VST MIDI Out - Recording" for each track VST plug-in or VSTi. This causes the MIDI data from the VSTi to be sent and recorded in another track.

By activating the option "VST MIDI Out - Recording" on a track (right-click on the record button), you can record all received VSTi MIDI Out data in the corresponding track. If MIDI Thru is activated, this signal can also be routed to another VSTi/VST FX or external MIDI device.

Permanent Options: These additional settings are valid for all instances of each plug-in.

Note: We only recommend changing the "Permanent Options" when compatibility problems occur. If you make changes here, reload the project to apply them.

Limit to only 1 CPU: All instances of a VSTi will be calculated only on one CPU. This avoids any multi-CPU conflicts between various instances.

If you apply this option to an effect plug-in, all track and object effects on the track that use this plug-in will now be calculated only on one CPU. As soon as you activate this option for an effects plug-in, the entire track and all its effects will be calculated only a single CPU.

Note: When using UAD cards, this option is activated by default.

Force calculation by silence input: When letting single tracks be processed as Economy tracks over the playback engine (view page 79), Samplitude normally doesn't process track signals if no audio signal is present. This decreases the load on the CPU.

With this plug-in option you can ensure that the entire track to which the plug-in is connected is calculated without interruption even when no audio signal is running through the effect. This is quite useful for delay plug-ins on Economy Tracks or for bouncing Economy Tracks, for example. This option is also recommended for plug-ins that produce their own audio regardless of the input.

Note: A continuous calculation, even of empty or silent tracks, takes place in the Hybrid Engine. This function can be deselected in the global Performance options (view page 609) (keyboard shortcut "Y") by checking the option "Deactivate FX on empty or silent tracks for ASIO".

No automatic copying: This option prevents automatic copying of plug-ins when you split or copy objects.

Load/Save patches/banks: Here you can save an instrument's settings and sounds. The standard formats for this are: *.fxp for patches or *.fxb for whole banks. Some instruments have their own patch/bank format. In this case, settings are usually loaded and saved directly through the instrument interface.

Set parameters randomly: Use this function to set all parameters of an instrument to a random value. For synthesizers this option can lead to surprising results and provides an interesting option for sound design thanks to the random generation of new sounds. However, please keep in mind that only the parameters that are accessible through the interface can be set randomly. With some very complex virtual synthesizers or modular systems, some parameters may not be adjustable using this option. It is important to note that due to the random placement of various parameters, very extreme sounds can be generated. This may lead to no sound being produced at all or extremely loud volumes and frequency ranges being reached. For this reason you should keep an eye on the monitoring volume while experimenting with this function.

Automate next parameter: The next parameter that you change during playback will be recorded with automation. You can also do this by holding "Ctrl + Alt" while clicking and changing the parameters with the mouse.

Hardware Controller Easy Learn: When you activate this function, you can set up your controller by

1. dragging and dropping the plug-in element that should be learned.
2. Move the element on the hardware controller you selected

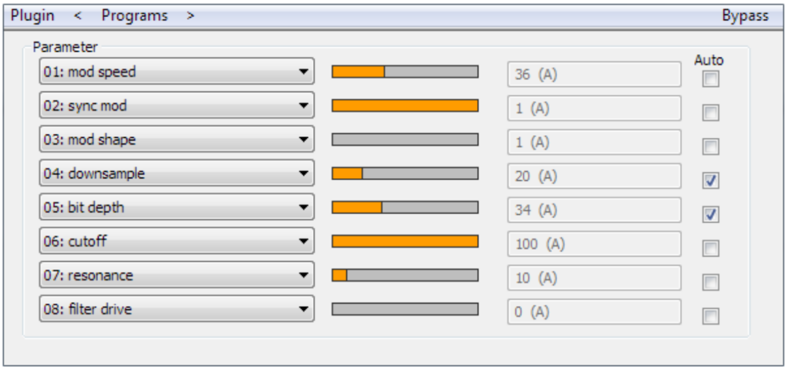
Pay attention to the following:

- When you are finished assigning the controls be sure to shut "Hardware Controller Easy Learn" off to avoid making any unwanted changes to the settings.
- The hardware controller element must first be learned in the hardware controller setup and cannot be left empty.
- Learned elements permanently modify the internal mode in the hardware controller setup. The assignment is also available later for other projects.
- It can be cleared again with the "Restore modes" button beside the internal mode in the hardware controller setup

> Detailed information about adding hardware controllers is provided under "Hardware controller (view page 476)"

Plug-in parameter dialog

If the selected plug-in does not have its own interface, the parameter dialog will open. This display mode is limited to just a few control elements and can only be opened via the menu of the instrument window. In this dialog, eight freely selectable instrument parameters are clearly and numerically displayed. Bar control elements may also be used to adjust the parameter values.



Once the automation of the selected parameters is activated, the automation curve will be displayed in the project track for editing. You can automate parameters during playback via the corresponding bar faders.

The parameter selection is saved for each plug-in so that the next time you use the same plug-in, the same parameters are used. This provides the advantage that every frequently used parameter of the corresponding instrument is available immediately after it has been opened.

Play and monitor instruments live

Monitoring

Use the following settings for live monitoring:

- ASIO driver ("File > Program Preferences > System/Audio > Audio system > Driver system")
- Activate global record monitoring in the transport control ("Moni" button)
- Activate the track monitoring button (loudspeaker symbol)
- Software monitoring or Mixer monitoring ("Menu, File > Program preferences > System/Audio > System options > Audio system > Monitoring Setup")

Tip: You can also select this monitoring mode by right clicking "Record monitoring" on the transport control.

Latency: Please note that a system-dependent delay between pressing the key and hearing the sound occurs when you play virtual software instruments. This so-called latency time is mainly determined by the buffer size set for the ASIO drivers. To play in an acceptable manner, we recommend a buffer size of max. 1024 samples. This corresponds to 23ms at 44.1kHz. For many users, however, a latency of 3 ms, i. e. 128 samples, is optimum. Please note that the CPU load also increases at a lower buffer size. The "live" delay only occurs when an instrument is played; playback latency of an already recorded MIDI track via an virtual instrument is automatically compensated by Samplitude.

As software instruments are fully integrated into the audio engine of the program, the signals can be routed, mixed and equipped with plug-ins any way you like. There are restrictions when recording VST instruments, according to the selected monitoring settings.

Note: For monitoring effects during input, "Track FX monitoring" mode or "Mixer FX monitoring/Hybrid Engine" must be activated.

Recording and playing back an instrument

An instrument is recorded in much the same way as a normal MIDI track. Make sure that the MIDI recording is active in the desired track and then press the record button in the transport control. The recording begins immediately.

When playing back already recorded MIDI tracks, track monitoring should be switched off.

To be able to record MIDI data sent by a VST instrument in another track, activate the "MIDI out" function in the plug-in panel menu for the respective track VST plug-in or VSTi. This sends MIDI data from the VSTi.

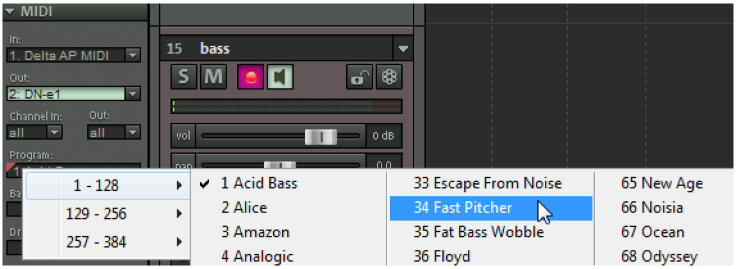
By activating the option "VST MIDI out recording" for a track (right click the record button), you can record all received VSTi MIDI out data in the corresponding track. Right click on "REC" in the track editor and set a check next to the corresponding context menu item.

You may record VSTi audio output for VSTi return tracks by activating recording in these tracks.

If "MIDI thru" is activated, this signal can also be routed to another VSTi/VST-FX or external MIDI device.

VST instrument preset selection

If an instrument has presets, these are displayed in the "Program slot" ("Prg") with their internal name (if "MIDI" mode is activated).



This dropdown menu can be used to easily "step through" the presets of your instrument. These presets may also be selected using the instrument window's "Program" menu.

If your MIDI input device can send "Program change" commands, you can also directly select programs of this instrument. The program slot is automatically updated in the track to provide an overview of the program at all times.

Sidechain Input

At the track and master level of a correspondingly equipped VST/MAGIX plug-in (e. g. am-munition), as well as in advanced dynamics, you can activate the sidechain and specify one or more of the previous tracks as the sidechain input(s). In the tracks or mixer channels sought out as side chain signals the track name of the fed track appears in the AUX section just above the send level.Internally, a side chain bus (AUX bus) is created.

Right clicking on the AUX send button selects this as "Side chain send".

The "Side Chain Solo" function bypasses the corresponding effect and only the input of the side chain is played. When closing the plug-in dialog, the "Side chain solo" function is reset.

The command "Sidechain filter" opens a parametric equalizer that may be used to edit the sidechain signal.

Detailed information on the sidechain function can be found in "Effects -> Dynamics -> Advanced Dynamics -> Dynamic parameters (view page 767)".

ReWire Client Applications

ReWire-compatible client applications (e. g. Propellerhead Reason) can be integrated into Samplitude as synthesizers.

Activate ReWire functionality in the System Options settings ("Y" key) under "Effects -> VST/ReWire". Afterwards, installed ReWire applications can be loaded as instruments in the MIDI out slot of Samplitude. 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 Samplitude and should be closed before exiting Samplitude. Several client applications can be opened automatically by right-clicking on their names in the MIDI out slot.

The ReWire application can be controlled with MIDI, just like a software instrument. Here you can channel the individual output signals of the ReWire clients to several tracks according to the routing for the multichannel software instruments. The client application runs, starts, and stops synchronous to the time position of Samplitude.

When using ReWire the "classic" MIDI channel of MIDI notes and events is not important and is replaced by the ReWire MIDI bus system. In this case the MIDI object of a track controls a ReWire MIDI bus. This means that all events of a MIDI object apply to the track for this ReWire bus, independent of the channel number the events originally had. Multi-timbral MIDI objects (e. g. 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 MIDI buses. A ReWire client only registers the MIDI busses that are actually available with the host (Samplitude). You can therefore select the bus within the track in Samplitude's MIDI channel menu (e.g. the receiving instrument for Reason).

Only a few ReWire clients can be opened directly through the host. If you start your client application manually, it will automatically detect a host application opens in a special Client Mode.

Automation

When mixer settings such as volume, panorama and effect setup for a project are modified to enable (for instance) the fading in of tracks, audio signal movement in panorama or panorama changes for effects, this is known as automation.

The display and drawing of this automation is done using automation curves. An unlimited number of automation curves are available on each track. You can draw automation movements on track and master level during playback in automation curves. All automation curves (track, master and object automation) can be edited after the fact or directly drawn in.

- The following dynamics automation settings are available at track level: volume, panorama, surround, AUX sends, EQs, and plug-in control elements and MIDI controllers.
- At object level, you can automate the volume, AUX sends, and VST plug-ins.
- At master level, you can automate the volume, EQs, VST plug-ins, and MIDI controllers.

Create automation curves

There are specific control elements on the track header for activating the most important track automation curves – volume and panorama.

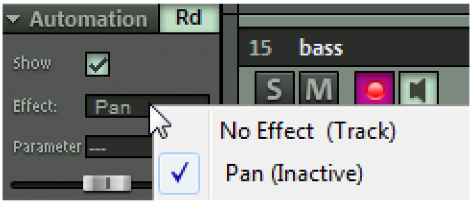
You can select the other parameters for automation in different ways:

- Select parameter in automation context menu or menu automation
- Via keyboard shortcut Ctrl + Alt + Control element in the mixer or the interface of a plug-in
- Via the VST parameter dialog

Automation context menu or menu automation

In the automation menu or automation context menu (right-click on "Effect" in section "Automation" in the track editor) you can select a automation parameter. Below in the menu, all available parameters are listed in sub-menus according to loaded plug-ins and effects. Select the parameter you want.

All available active or inactive automation parameters are listed in the top part of the menu. Here, select an automation parameter that you can then manipulate during playback with the automation controller in the track editor. The curve of the selected parameter can be seen in the track in the foreground. Via the context menu, you can opt to have all other curves displayed as thin lines or you can also have them faded out (See Automation – Context menu (view page 443) for more details).



Ctrl + Alt + Control element

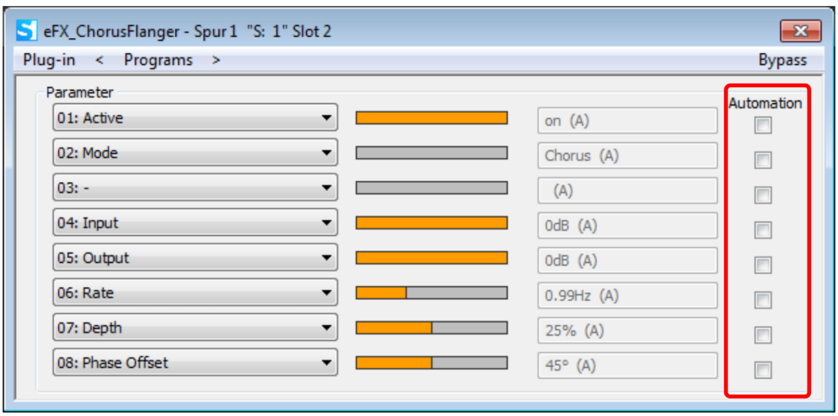
This is the fastest and smoothest way to automate a plug-in parameter. In the mixer (volume, pan, aux sends, EQ) or interface of a plug-in, click the control element of the parameter you want to use while holding down Ctrl + Alt. In stopped state, this then creates a curve. During playback, a change to the parameter is drawn directly (regardless of the selected automation mode (view page 430)).

Note: This works only at track level, and not for the automation of object effects (view page 437).

An alternative method of automating a parameter is to select "Automate next parameter" in the plug-in menu of the VST plug-in dialog and then click the corresponding control element.

Automation via VST plug-in/VST parameter dialog

You can also select and activate all parameters you would like to automate in the parameter dialog of the VST plug-in. You can open the parameter dialog in the plug-in's menu via "Plug-in" > "Parameter dialog".



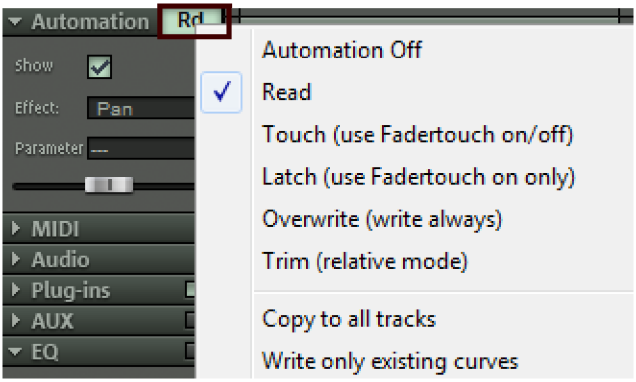
Select the parameter you want from the drop-down menu at the front (the first eight parameters are selected by default) and activate the box under "Automation".

Tip: To display the new parameter in an Automation Lane, select "Lanes for all curves" from the Automation menu or the Automation context menu.

Detailed information about the VSTi parameter dialog is provided in the chapter "Plug-in parameter dialog (view page 424)".

Automation modes

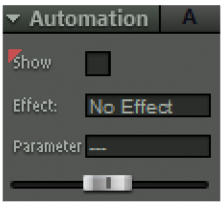
You can draw automation using various modes.



The automation modes define how automation data should be written or played back. You can set the automation mode for each channel/track. Various automation modes may be selected by right clicking the automation button in the mixer channel or track editor. Clicking this button toggles between the last activated mode and "Read" mode.

Off mode

All automation functions are deactivated in "Off" mode, all previously written automation sequences are ignored.



Read mode

All recorded automation parameters are played in "Read" mode. The automation button is usually set to this state.

Touch mode

In "Touch" mode, automation data is only recorded as long as you touch the selected control element with the mouse or touch it from your external controller. As soon as you release the control, the automation recording stops. Once released, the control will move back to the previously recorded position. You can determine the fader return time in the System Options (Keyboard shortcut: „Y“) under „Effects“ > „Automation“ > „Automation Release Time“.

Latch mode

In "Latch" mode, automation files are written starting only from the first touch of the control element or the external controller. As soon as the corresponding button is released, the automation is written to the last value until playback ends or another mode is switched on.

Overwrite mode

In this mode, automation data is recorded immediately as soon as playback begins, irrespective of whether you touch the fader. The automation continues to be written until you end playback or switch into another mode. This allows you to quickly overwrite a previously recorded automation curve.

This mode only affects the previously selected curve, regardless of whether other curves are active, since overwriting all automations at the same time is not useful.

Trim Mode During Playback

"Trim" mode produces a relative change of values during simultaneous playback for existing volume automation.

Note: For other automation parameters, trim mode during real-time drawing behaves exactly like touch mode (view page 431).

In Trim mode you can move pre-existing automation data. The channel's volume fader will initially be set to 0dB, regardless of the value of the current automation data. The fader display that shows the value relative to the changes (fader value box) will show the output value as -0.0 dB.

This mode is useful if you want to maintain the existing automation movements, or increase or decrease the overall level.

When trimming a range, new automation points will be set at the range edges. If you move the fader for a relative volume adjustment during playback, the relative adjustments will be displayed. You can see the current difference to the previously recorded value in the fader value box as well as in the tool tip which appears.

If the option "Use snap for automation curve points" (System Options -> Program -> General) is selected, automation points can be set while moving set values on the grid.

This mode is useful if you want to maintain the existing automation movements, or increase or decrease the overall level.

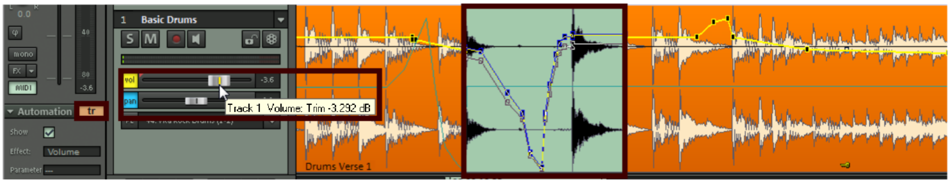
In "Trim" mode, automation data is only recorded as long as you touch the selected control element with the mouse or touch it from your external controller. As soon as you release the button, the automation recording stops. Once released, the control element will move back to the previously recorded position.

You can determine the fader return time in the System Options (Keyboard shortcut: „Y“) under „Effects“ > „Automation“ > „Automation Release Time“.

Trim mode with stopped playback

You can also make relative changes to all volume automation points within a certain area when playback has been stopped.

Select a range for the respective volume events in the track and drag the channel fader in "Trim" mode up or down. You can see the level adjustment difference value in the Tooltip or Fader box.



Note: You can of course activate any number of automation points for all other Automation curves (e. g. EQ, Panorama etc), by clicking whilst holding the Shift key, then change the values of the selected Automation points, again by holding Shift.

Apply to all tracks

"Copy to all tracks" applies the currently selected automation mode to all tracks/channels.

Automate existing curves only

Select the option "Automate existing curves only" to apply the automation data to existing automation curves only. This enables you to experiment with parameters that have not been recorded yet, without having to write automation for these parameters directly. This option should be removed once the new parameters are ready to be committed.

Preview automation (all tracks)

This option temporarily overrides all write automations.

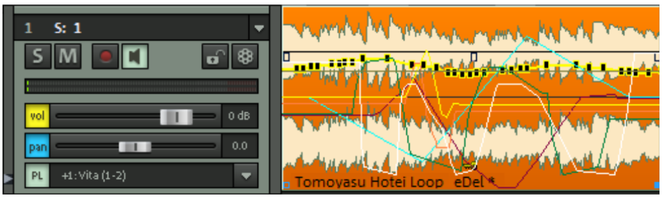
The background is as follows: You have a project that already contains automations. Now you want to add a new section at the end of the project and make changes to the mix there. This creates a problem: When a track is in Read mode, you cannot change an existing parameter because it is controlled by the automation curve. With a writing automation mode (Touch, Latch, Overwrite), new automations are written immediately and transitions are created between the new parameter values and the existing ones, which you may not want. In addition, new curves are created for parameters that were previously not automated, which means that these parameters are also changed in the existing section of the project.

Follow these steps to resolve this issue:

1. Activate a writing automation mode (Touch, Latch, Overwrite)
2. Activate the option **Preview automation**.
3. Start the playback and make your changes in the mix. Existing automations are not changed. However, if you change a parameter that was not previously automated, a new automation lane is created that retains the existing value (not the changed one!).
4. Stop playback, the play cursor jumps back to the beginning of the new section.
5. Then choose **Set parameter value(s)** from the Automation menu and one of the options to create jumps or transitions or use one of the buttons in the Automation Panel (view page 436).
6. Depending on the option selected, jumps or transitions from the existing to the new automation values are inserted at the cursor position.
7. You can now deactivate the option **Preview automation** again and continue working with the automation modes as normal.

Automation lanes

When several automation curves are present on a single track, things can become cluttered quickly and editing curves is more complicated, e.g. it is more difficult to select a curve when drawing one.

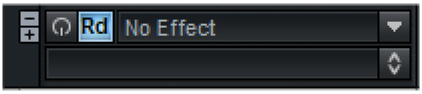


Several automation curves over one object

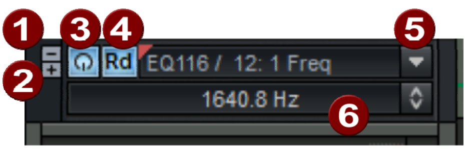
Automation lanes provide a solution for this issue. Automation lanes enable users to display an automation curve in a separate track, which make it easier to see and edit the curves.

You can use the little triangle below left on the track header to open and close automation lanes.

When you open the lanes for the first time, an empty automation lane will appear.



Click on the triangular button on the right to open a menu where in the section above, you can choose an automation curve to display in the automation lane. In the bottom section of the menu, you can also select any other parameter to create a new automation curve that will be displayed in the lane.



1. Clicking on the "**-**" key automatically fades out automation lanes.
2. Clicking on the "**+**" key creates a new lane under the automation lane.
3. **Automation on/off:** Used to activate/deactivate the respective curve.

4 **Automation mode:** The automation mode for the corresponding track

5 **Parameter menu:**

6 **Parameter field:** The current parameter value will be displayed at the play cursor position. 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.

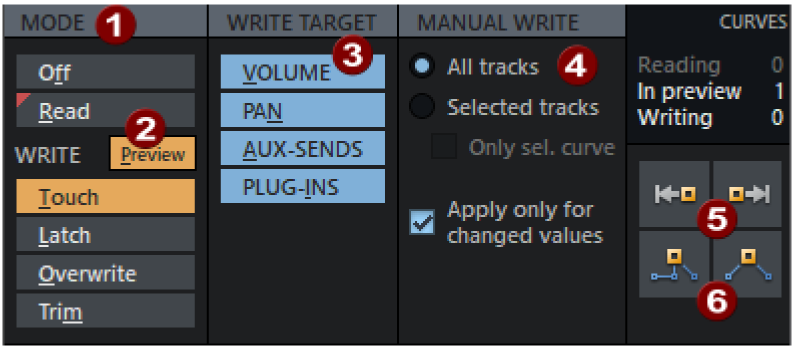
Tip: To quickly display automation lanes for all available automation curves, select "Automation" > "Lanes for all curves".

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 control 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 (view page 430) and Preview automation (view page 433)!



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 **WRITE TARGET:** 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 deactivated in Write Target act, irrespective of the selected write automation mode, as if the associated curve were being read in Read 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. 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 deviates 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.

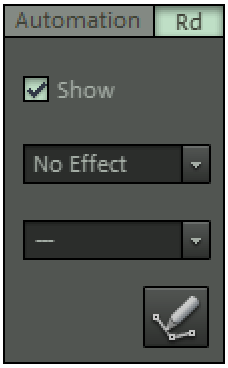
> These functions are only available in the Preview mode (Preview automation) (view page 433). For more information, read the corresponding section.

Object automation

At object level you can automate the volume, AUX sends, and VST plug-in parameters.

> Note: Drawing object automation during playback like such as with the tracks in not possible. You can only create automations as curves and edit these curves.

For automating object parameters, select an object and open the object editor (view page 146).



In the **FX** tab, click "Display" in the section "Automation".

Automation lanes are opened and existing track automation curves are displayed in automation lanes, which frees up space for drawing over the objects in the track.

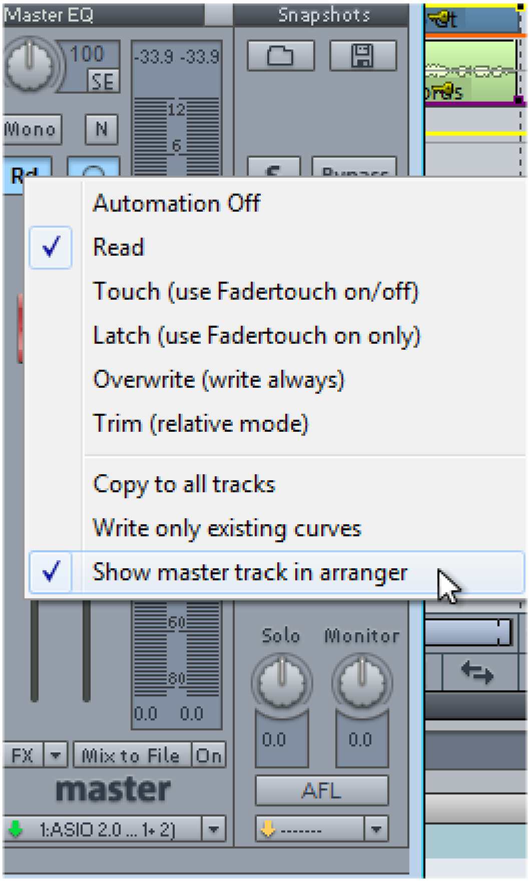
Click on the arrow button on the drop-down menu and search for the parameters you want to automate in the bottom section of the context menu.

Switch to automation draw mode and draw the parameter changes directly in the object.

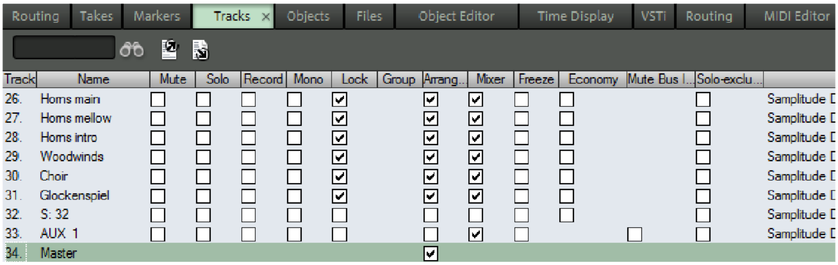
To transfer volume and panorama automation from a track to an object, select the option "Automation" > "Edit Selected Curve" > "Transfer Track Automation to Object". You can reset the object automation to track automation at any time using the command "Switch object automation to track automation".

Master automation

Volume and effect settings for the master track can be automated. To edit these track curves, you must display the master track in the arranger. Right-click the automation button in the master channel and select the option "Display master track in arranger". In the arranger, an additional track with the name "Master" appears, which displays the master automation track curves.



You can also make the master track visible in the arranger window via the track manager (view page 182). To do this, open the track manager and place a check next to "Arrangement" for the "Master" track name.



Recording automation in "Read" mode

The easiest way to write automation in "Read" mode during playback is via the "Ctrl + Alt + switch/controller" shortcut, while working directly in the plug-in/instrument/mixer interface. This way allows an automated recording to be made at any time. As long this key combination is held down, you may also automate the selected parameters in "Read" mode.

If a mixer/VST control element is clicked with the same key combination "Ctrl + Alt" while stopped, a corresponding inactive curve will be created automatically for editing in "Draw" mode.

Editing Automation Curves

There are various options for editing curves:

- In universal mouse mode, you can create and edit individual curve points.
- With curve mode or combined object and curve modes, you can edit curves in detail
- With automation draw mode you can freely draw curves

Editing automation curves in universal mouse mode

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

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

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

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

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

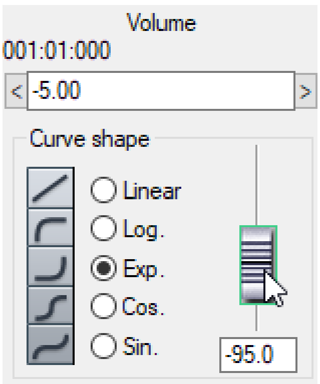
Moving curve segments: If you move the mouse pointer over a curve (the mouse pointer becomes a double arrow) and drag on the curve, both handles to the right and left of the mouse are selected and moved together.

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

Automation points lasso: Clicking and dragging next to the object activates the object lasso (view page 104) for selecting multiple automation points.

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

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

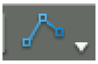


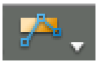
Close the entry field by pressing the Enter key. When the input field is open, you can jump to the next automation point using the Tab key.

Under Waveform you can define how the curve runs from the selected curve point to the next. The possible curve shapes correspond to the curves for objects fade in and out (view page 153). Using the fader you can adjust the precise settings of the curve form of the fade.

Object/Curve mode

The special curve mouse modes enable more targeted editing of curves. Curve points can be set with a simple click and selecting curves with lasso is easier, since range and object functions do not include universal mode.

 Curve mode

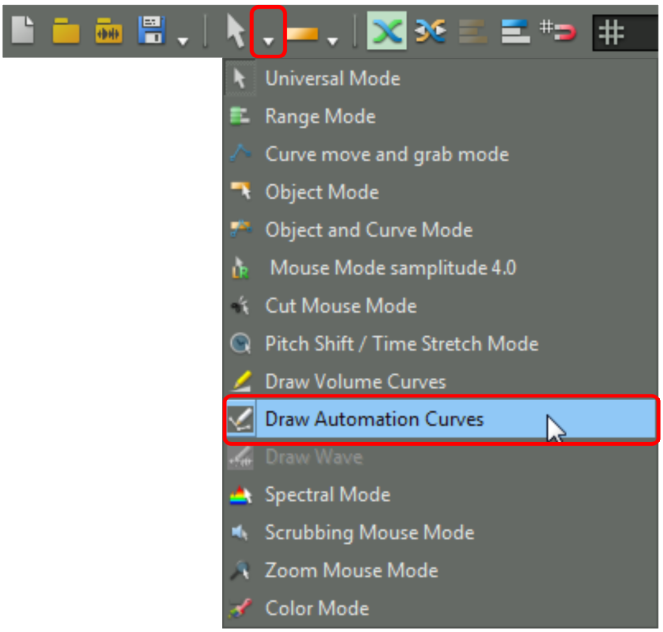
 Object/Curve mode

In curve mode you can easily create new points on the curve, move curves and delete them with a double-click. If your mouse is not directly above the curve, you can select an area of automation points using the lasso.

In Object/Curve mode you can also create new points on the curve, move, and delete curves with a double-click. If you move the mouse pointer away from the curve, you will automatically be in Object mode. In this mode you can move objects and adjust their start and end positions, add fade-ins and fade-outs as well as edit the object volume.

Automation draw mode

Select automation draw mode in the toolbar to draw automation curves for a parameter or a MIDI controller.



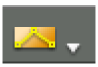
You can edit an automation curve for the selected parameter using the pencil tool. Simply click anywhere in the track display and then draw the automation curve by dragging with the mouse cursor.

If you press the Shift key while drawing, a curve will not be drawn. Instead, this enables you to drag an individual curve point to the desired location and the curve progresses in a straight line from the previous curve point.

VST parameters are shown in % while drawing. Volume and AUX send parameters are indicated in dB.

Move automation curve with audio/MIDI data

To move the audio material to a new position on the timeline along with the automation curve, the automation curve will have to be connected to the object. This is possible via "Link curves to objects" mouse mode. Click this button in the mouse mode toolbar to get started.



Objects may now be moved or copied together with your curve points.

Note: If you only want to move or copy curve points, follow the instructions above and simply delete the objects once they have been moved/copied. The curves remain intact.

To copy the curve points, simply use "Curve edit" mode and drag out a section by moving the mouse from right to left over a curve while holding down the mouse button, copy the curve points via "Ctrl + C", and then insert them into a different track at the current play cursor position via "Ctrl + V".

Automation – context menu

A number of options are available to open the automation context menu:

- Access the "Automation" menu item
- Right click the "Automation" button in the track editor
- Right click the automation controller in the track editor
- Right click the "Vol" button in the track box

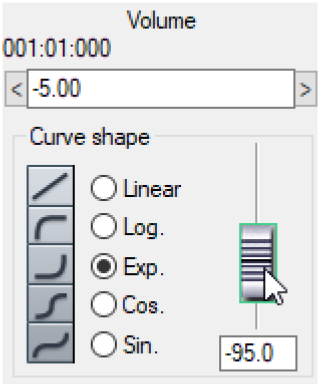
The following functions are now available:

The first entry indicates which parameter has been automated, e.g. "**Volume**" or "**Pan**". Checking the corresponding entry activates the corresponding automation curve. If an element hasn't been automated, then "**No effect (track)**" appears.

Editing curves

Create new point at playback position

A new curve point will be added to the curve at the play cursor position, and the input field for numeric data entry opens.



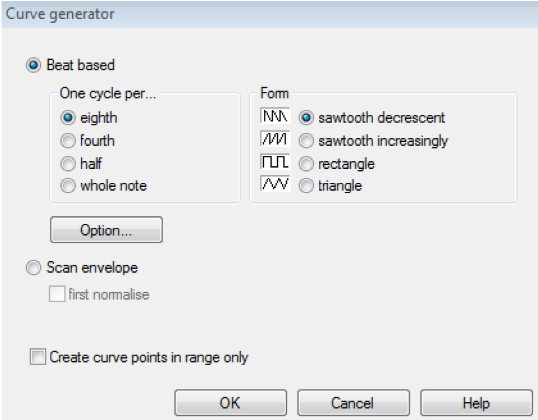
Edit next point at playback position

The input field for numeric data entry will be opened from the play cursor position at the next curve point.

Curve generator

The curve generator will create a volume curve for you that is beat based and pulses according to eighth, quarter, half or full notes. You can enter a minimum and maximum value or define delay values as an option for the beat-based envelope curve calculation.

The option "Create..." in the automation context menu opens a selection dialog to define the shape of the automation curve more accurately.

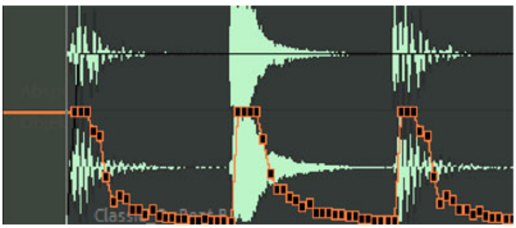


Beat-based: This option causes the form of the automation curve to follow the beat in the arrangement.

There are 4 basic patterns available for beat-based automation curves: 2 saw-tooth shapes, rectangle, and triangle. These shapes run depending on the beat and control the intensity of the activated effect. The left side of the dialog enables the automation curve to be set to activate once per eighth, quarter, half, or whole note.

Options: This opens an additional dialog to control the style and intensity of the influence of the beat on the automation curve (see below).

Scan envelope curve: Allows the volume process to be displayed as an automation curve as an alternative to a beat-based automation curve.

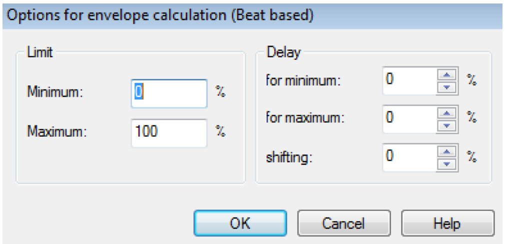


First normalize: This function normalizes the level of the audio object before the object is scanned to create an automation curve.

Create curve points in range only: If this option is selected, curve points will be generated only in the area previously defined in the arranger.

Curve generator - options for envelope calculation (beat-based)

Limitation: Determine the minimum and maximum value of the envelope. Via Delay Minimum/Maximum the basic forms can be altered further.



Shift moves the entire curve. This lets you create interesting off-beat effects.

Edit curve – invert

This command inverts the activated automation curves.

Edit selected curves – thin out

While recording, the automation events are placed in very short intervals. The command "Thin out" reduces the number of curve points. The course of the automation is then displayed and reproduced even more accurately.

Thin out automatically

The command "Thin out" is executed automatically after automation data is recorded or drawn.

Edit curve – transfer track automation to object

You can use this function to transfer volume and panorama automation to objects. If the track in question already has automations, you will be asked if you want to overwrite existing track automations or overlay them with a new curve.

Edit curve – transform object into track automation

You can use this function to transfer volume and panorama automation from objects to tracks.

If the track in question already has automations, you will be asked if you want to overwrite existing track automations or overlay them with a new curve.

Edit curve – inactive

Use this command to deactivate the selected automation curve. The curve remains intact, but will be displayed as a dotted line and will no longer affect the corresponding parameter.

Edit curve – curve color

This command displays a color palette for you to specify the color of the active curve.

Logarithmic

This option lets you switch volume curves to logarithmic display.

Edit curve – copy, paste, delete

The activate automation curve may also be copied, pasted into other tracks, or deleted.

If the option "Use snap for automation curve points" (System options -> Program -> General) is selected, automation points may be set while moving set values on the grid.

Delete All Curves

This command deletes all automation curves in the corresponding track.

Automation Mode

Specify the Automation mode here.

You can find detailed information in "Automation modes (view page 430)".

Set Parameter value(s)

This command creates new automation points in all active and inactive automation curves of the selected or all tracks according to the currently set parameter. This allows, in connection with the option **Automation Preview (view page 433)** , to explicitly write the automation of the entire project at a certain point, in order to prevent existing parameters from being changed by new automations and unwanted transitions from occurring. The following options are available:

- **Jump to current value:** New points are inserted as jumps in all automation curves.
- **Jump to current value (selected tracks):** New points are inserted as jumps in all automation curves of the selected track(s).

- **Glide to current value:** New points are inserted in all automation curves as a transition to the previous value.
- **Glide to current value (panorama only):** A new point is inserted in the panorama curve of the selected track as a transition to the previous value.
- **Glide to current value (selected track):** New points are inserted in all automation curves of the selected track as a transition to the previous value.

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!

Display automation curves options

Do not display automation

With this option, you can deactivate the display of the automation curves for all tracks.

Display track automation (default setting)

If you select this display option, only the track automation will be displayed.

Show lanes and object automation

If you select this display option, only the object automation will be displayed for all tracks. Simultaneously, this option opens the automation lanes and displays the existing track curves in the lanes.

Show Only Selected Curve

Views only the selected curves for the corresponding track. This helps to maintain a clear overview if several automation curves have been created.

Show unselected curves (not selectable)

The unselected automation curves are also shaded, but may now be activated with the mouse.

Show all curves (selectable)

The deselected curves are also shaded, but may now be activated with the mouse.

Show lanes for all curves

This opens the automation lanes and corresponding automation lanes are created for all existing automation curves.

Select previous curve

With this you can select the last edited curve.

Select Next Curve

With this you can call up the next curve from the selection list.

The last section of the context menu lists the ranges for which automation parameters are available during automation of the corresponding track.

Synchronization

Synchronization is very important to an audio system, since it is necessary to align playback and recording speeds of different system components. Synchronization enables studio equipment such as tape machines, drum computers, video recorders, or sequencers to be linked with different formats like MIDI clock, MIDI time code, or SMPTE time code so that all of the devices run at precisely the same time.

Samplitude can be configured to as the master or slave within a synchronization system. As a master program, it generates the necessary timing information required for other system components; as slave. Samplitude receives the time code data to follow during playback or recording.

Clock on digital systems

In case digital signal flows are processed, e.g. via ADAT, SPDIF, or MADI, it is important to define a clock reference within the overall system. The clock signal is transferred via Blackburst, World Clock, or the digital input, and it may be received by the receiving or playback computer. External devices such as converters or mixers may also provide a clock reference. For these systems, there can be only one master but several slaves.

Note: Don't confuse (Word) clock with time code. The clock signal is only a digital pulse for comparison of the timing between the connected devices and to keep the bit rate constant. This enables transmission errors to be prevented. Professional digital audio devices normally feature a word clock input and are able to generate and receive a clock signal.

Time-related synchronization of projects

For more complicated configurations, it is usually necessary to connect various audio and video systems via absolute time references. This enables them to operate according to the behavior of the master, and the time code information (transport and progress) is implemented correspondingly.

Synchronization formats

The synchronization format features slave device information about the start position, start and stop signals and, in extended formats, the precise timing information, which are continuously transmitted.

Samplitude understands and transmits the synchronization formats MIDI clock, MIDI timecode (MTC), and SMPTE (Society of Motion Picture and Television Engineers).

MIDI Clock

The MIDI clock acts as a MIDI transmission clock consisting of system messages (F8H), and its time reference is not based on hours, minutes, and seconds, but rather on "ticks". One "tick" is the 24th part of a quarter note (24 ppqn [pulses per quarter note]) – MIDI clock signals are therefore transmitted 24 times per quarter note. Since the MIDI clock is based on the tempo as a reference, the number of ticks output depends on the tempo of the generating device. The MIDI clock also does not transport any information concerning where in the song the listener is located at any particular time.

Depending on the options the external MIDI devices provide for synchronization, start (FAH), stop (FCH), and continue (FBH) signals are also transmitted. These parameters ensure that the connected devices start via MIDI, stop at any point in the song, and that they may be restarted at the stop position.

Depending on the device specification, the MIDI clock signal may also transmit the song position pointer (SPP). This acts as a beat counter that provides the number of MIDI beats that have been played back since the start. SPP is also sent all 6 ticks as system exclusive data, which corresponds with a 1/16 note rhythm.

The song position pointer makes it possible to link the connected devices with a precision of 1/16 notes at any position in the song.

Note: Look at the specifications of your external MIDI devices to learn which synchronization parameters are supported. Samplitude is able to process MIDI clock (F8H), start (FAH), stop (FCH), continue (FBH), and song position pointer (SPP).

MIDI clock synchronization is especially useful for synchronizing devices with ticks as a time reference with Samplitude, e.g. external samplers or drum computer or if you want to link LFOs or arpeggiators of multiple external synthesizers.

SMPTE Timecode

SMPTE Timecode is a conventional synchronization standard for linking different audio and imaging devices via absolute time reference. Every SMPTE message sends exactly how much time has passed since the beginning of the projects.

The SMPTE Timecode refers to the units hours:minutes:seconds:frames. The concept of "frames" originates from the film and video industry, which the SMPTE code was originally developed for. The different frame rates specify the resolution in the individual images per second, which is common in the film and video industry.

MIDI Time Code (MTC)

MIDI Time Code (MTC) is an eight-digit format like SMPTE timecode that integrates an absolute timecode (hours, minutes, seconds, and frames) as a synchronization reference. MTC is transferred with other MIDI data via the MIDI interface and transmitted as system messages or as universal system exclusive messages.

MIDI Time Code is especially useful for synchronization of multiple sequencers or DAWs.

Note: To synchronize devices that output SMPTE Timecode with MTC, the SMPTE Code is converted into MIDI Time Code first via an SMPTE to MTC Converter.

MIDI Machine Control (MMC)

This is a signal for controlling external controllers that support the MMC format. Samplitude may also be remotely controlled via MMC commands from an external controller or mixer.

You can find out more about this is available under MIDI Machine Control (MMC) (view page 456).

APP (ASIO Positioning Protocol)

The ASIO Positioning Protocol enables transmission of sample-exact time information from a third device that is connected via a digital interface with the sound card. As a component of the ASIO driver architecture, digital signal connections (SPDIF, ADAT) or special sound card interfaces (LTC, Video Burst In) may also be used to transmit the ASIO Positioning Protocol. This enables synchronization with SMPTE or MTC-equivalent timecode, without requiring an additional MIDI connection. Samplitude is therefore able to follow the input signal of an external source exactly.

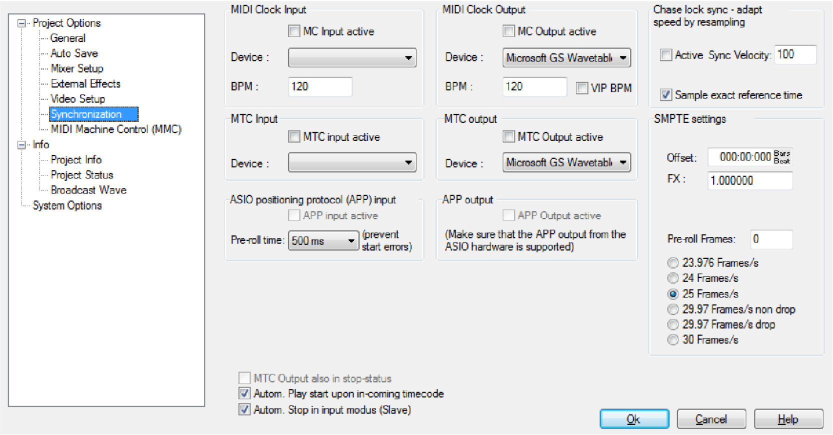
Note: Inform yourself regarding about whether your audio device and the ASIO driver support the APP protocol.

Synchronization Setup Dialog

The settings options for synchronization are available in the synchronization setup dialog.

Menu: „File“ > „Program Preferences“ > „Synchronization Setup...“

Keyboard shortcut: Shift + G



This window is for customizing the synchronization according to your requirements. The available formats are MIDI clock, MTC, APP, and SMPTE.

MIDI Clock

MIDI Clock Input

MC input active: Activate the input for the incoming MIDI clock signal by clicking this check box.

Device: Select the device that should create the MIDI clock signal for synchronization.

BPM: Define the tempo that the MIDI clock should receive.

MIDI Clock Output

MC output active: This check box activates the output for the MIDI clock signal generated by Samplitude.

Device: Select the device that should receive the MIDI clock signal for synchronization.

BPM: Define the tempo that the MIDI clock should receive.

VIP BPM: This check box defines the project speed set in Samplitude for the synchronization project.

MTC (MIDI Timecode)

MTC Input

MTC input active: This check box activates the input for the incoming MIDI Timecode signal.

Device: Select the device that should create the MIDI timecode signal for synchronization.

MTC output

MTC output active: This check box activates the output for the MIDI timecode signal generated by Samplitude.

Device: Select the device that the MIDI timecode signal should be sent to for synchronization.

ASIO positioning protocol (APP) input

APP Input

APP input active: This check box activates the input for the incoming APP input signal.

Preroll time: In order to hinder problems during starting, you can set the preroll time at up to 1500 milliseconds.

APP output

APP output active: This check box activates the output for the APP signal generated by Samplitude. Please make sure that the APP output from the ASIO hardware is supported.

> Information concerning the generation of SMPTE timecodes within Samplitude is presented in the menu item "Effects -> SMPTE generator (view page 980)".

Automatic Stop in Input Mode (Slave)

If Samplitude is receiving MTC from an external device and the corresponding track is ready to record, Samplitude will begin recording as soon as the external signal is recorded. If sending of external synchronization signals stops, Samplitude will end recording and enter into playback mode automatically. If the timecode is received again, Samplitude will begin playback. If you want Samplitude to record again when receiving MTC, activate this option.

Automatic Playback with Incoming Timecode

If Samplitude is receiving MTC from an external device and the corresponding track is ready to record, Samplitude will begin recording as soon as the external signal is received.

MTC Output also in stop-status

If you select this option, Samplitude continuously sends the current time position as MTC – even in case playback has stopped.

Chase lock sync - adapt speed by resampling

Samplitude supports real chase lock synchronization, i. e. audio playback can be controlled by an incoming timecode signal. In this case not only the starting point of audio playback is controlled externally, but sync also controls playback speed if "Chase Lock" is activated. Samplitude is therefore capable of following analog tape devices or VCRs that always feature some slip over longer periods.

If fluctuations in speed occur, Samplitude can make adjustments in slave mode that ensure exact temporal synchronization. This function is called "**chase lock**". Use chase lock in case a device involved in the synchronization cannot be clocked centrally via Blackburst, Wordclock, or the digital input and Samplitude is the "slave". This is the case if the timecode is located on a track featuring a multitrack machine.

Active: Activate chase lock synchronization by clicking this check box.

Sync Velocity: By entering the Sync Velocity you can influence the speed of the tempo adjustment. The greater the sync velocity value, the faster Samplitude will follow master device tempo changes; however, this also means that the pitch fluctuation in the audio material is greater. The preset value for the sync velocity is 100; experiment with values larger than 100 in case the synchronization does not run exactly.

Note: Please note that Samplitude will apply resampling to the recording in real time according to the timecode fluctuations in case chase lock is active. This causes increased CPU load and may result in unwanted changes to the audio material, if this is played later with other timecode references.

If the systems are coupled via a central digital audio clock, then chase lock should not be activated.

When using the **hardware pitching** feature, chase lock synchronization uses direct access to the sample rate of the sound card in 1 Hz steps. This achieves particularly precise synchronization without additional CPU load. However, this feature has to be supported specially by the sound card.

Use sample-accurate position: If this option is active, Samplitude uses the time position of the sound card as the clock reference (timer) and not its own internal timer. This ensures sample-exact synchronization between the recorded audio material and the sync signal.

SMPTE settings

SMPTE-Offset: Specify an offset to be deducted from the incoming SMPTE time before the time is used for synchronization. An offset of **01:00:00:00** (1 hour) therefore allows you to synchronize a tape with an SMPTE code that starts at 1 hour. The SMPTE offset behaves relative to the project start time (view page 84).

FX: This factor allows you to equalize possible irregularities with regard to positioning in long samples. Perfect synchronization at the beginning of the sample is nevertheless a prerequisite.

Preroll frames: Enter the number of frames that Samplitude should ignore before synchronization begins. This takes into account the fact that analog devices normally require a certain amount of time before they reach the correct speed. Samplitude will now synchronize at a valid time, since this startup time is able to be skipped according to the set preroll frames.

Type: Select a suitable frame rate, e. g. 24 for video/movies, 25 for PAL video and audio synchronization, 29.97 for NTSC video, and 30 for HDTV.

Synchronization - Samplitude as master

Samplitude supports MIDI Clock, MIDI Timecode, and APP Timecode output as a master.

Sync output is directly linked to audio playback so that no delays occur between Samplitude and external devices, even with long tracks.

Note regarding linking Samplitude as the master:

- In this case, use a virtual MIDI router for linking the programs internally.
- Wherever possible, favor the MIDI Timecode synchronization with MIDI Clock, since this does not need to include tempo changes.
- If Samplitude is running as the master, then set the FX factor to 1.0.
- For optimal synchronization stability, deactivate the virtual memory (if possible).

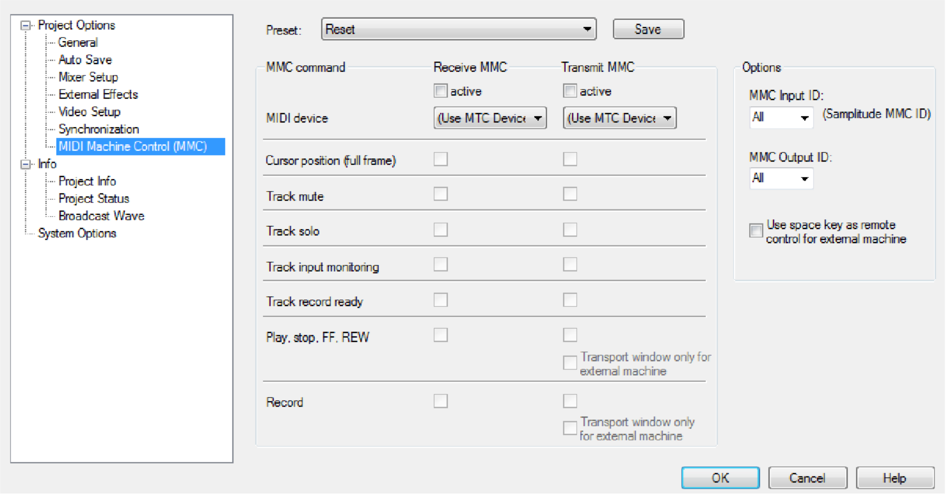
MIDI Machine Control (MMC)

MMC modes

In the project options, you will find a dialog for remote control via MIDI machine control. Samplitude supports synchronization of external devices via MMC. The following modes are available:

Receive MMC: If you have set up Samplitude to work as a slave, activate this option. Samplitude follows JOG shuttle operations, cursor position, track mute, track solo, track input monitoring, track record ready, play, stop, fast forward, rewind, and record commands from external devices.

Send MMC: Samplitude operates as master. When you use commands like cursor position, track mute, track solo, track input monitoring, track record ready, play, stop, fast forward, rewind, and record, the external device follows. If you also activate the "Receive MMC commands" option, Samplitude receives the current playback position of the device and displays it as an additional blue cursor on the timeline.



While playing a range in Samplitude, the MMC device stops once the end of the range is reached.

Transport window only controls external device

If you activate this option, the transport console no longer controls playback and recording directly within the program but instead controls the external MMC device. Play, stop, fast-forward, fast-backward and rewind no longer have a direct effect on Samplitude playback. The "Record" button starts recording in Samplitude in synchronization with the playback of the MMC device.

Use spacebar to control external devices: If this option is activated, the MMC device's start and stop commands are controlled with the spacebar.

Additional options

Input MIDI device: Set the MMC device which Samplitude receives.

Output MIDI device: Set the MMC device which Samplitude sends to.

MMC input ID: Enter the Samplitude MMC input ID here.

MMC output ID: Set the input ID of the MMC device you want to control remotely.

List of implemented MMC commands

Command	Receive	Send
Full frame	Yes	Yes
Stop	01	01
Play	02	02
Deferred play	03	-
FFWD	04	-
Rewind	05	-
Record ON	06	06
Record OFF	07	07
Record Pause	08	-
Pause	09	-
Locate	44	44
Variable Play	45	45
Shuttle	47	-
READ location	42 01	-
READ motion control tally	42 48	-
READ record status	42 4d	-
READ track record status	42 4e	-
Track Record Ready	42 4f	40 4f
Track Input Monitoring	42 53	40 53

Track Mute	42 62	40 62
Track Solo	42 77	40 77

MAGIX MMC extension

F0 7F <device_id> 06 6F <...> F7

Set Marker	6F 01	New auto numbered marker
Navigate Playback Marker to marker left/right	6F 02 n	n = 00 to left marker n = 01 to right marker
Set all tracks RecRdy on/off	6F 03 n	n = 00 off n = 01 on
Set Monitoring Mode	6F 04 n	n = 00 manual monitoring n = 01 tape monitoring
Set Input Monitoring on/off	6F 05 n	n = 00 off n = 01 on
Set Sync Mode on/off	6F 06 n	n = 00 off n = 01 on
Set Loop Mode on/off	6F 07 n	n = 00 off n = 01 on
Set Punch Mode on/off	6F 08 n	n = 00 off n = 01 on
Delete Punch Marker	6F 09	

Synchronization applications

Application 1. Synchronization using an external hardware sequencer. Samplitude is the master.

To synchronize Samplitude with an external hardware sequencer (drum computer, groove box, etc.), first select the type of synchronization. As these external devices usually use the song's tempo or rhythm-dependent information, they are often synchronized using MIDI clock. In this case, impulses are passed on from the master to the slave. These control impulses are transmitted for each quarter note. Their number per quarter is always identical. The tempo information is always clear, since the impulses do not occur in absolute time units but are rhythm-dependent. This secures the tempo synchronization of the two units that are to be aligned. Of course the same tempo (more or less) should be set to both units from the beginning. The MIDI song position pointer is another important MIDI clock synchronization tool which provides information on how may control impulses have already been transmitted. You can therefore start at any position of the sequencer. The same position of both synchronization units is thereby guaranteed.

Important: The value of the MIDI song position pointer is limited to 1024 bars (4/4 beat). After this, no further synchronization via MIDI clock is possible; connected slave devices simply stop. In this case, connected slave devices remain still.

1. Make sure that your external hardware sequencer supports synchronization via MIDI clock as slave.
2. Connect MIDI OUT of the master (MIDI OUT on the MIDI interface of the computer running Samplitude) to the MIDI IN of the external hardware sequencer.
3. Set the hardware sequencer to "Slave" mode. If you apply this setting, refer to the instruction manual for the respective device. You will now notice that the transport control and tempo setting of the hardware sequencer are deactivated. It will now receive data from the master, i.e. from Samplitude.
4. Open the synchronization window in Samplitude and activate the "MIDI clock output -> MC output active" option.
5. Select the MIDI port that connected to the slave device. Select the tempo you would like the external hardware sequencer to run at. This does not necessarily have to be the tempo of your Samplitude VIP project (VIP BPM), but in most cases it is sensible, since the bars in the Samplitude arranger window will correspond with the bars of the hardware sequencer.
6. If you press play in Samplitude, the external device runs in sync with the tempo of your VIP project.

Note: Please also observe that the "Transport window only controls external devices" option must be deactivated if the device that is to be controlled cannot process MIDI machine control data (MMC), since the transport keys neither control Samplitude nor the external sequencer.

Application 2: MIDI Clock synchronization using an external device. Samplitude is the slave.

In principle, it makes sense to choose a master within a group of devices with the most stable timing. In most cases, this would be Samplitude. Of course you can also set Samplitude as slave, for example, if the main parts of a production are prepared using an external hardware MIDI sequencer and Samplitude is only used as an additional audio device, or if the sequencer only permits MIDI Clock synchronization as a master.

The following should be observed: unlike a MIDI-only device, Samplitude can contain audio files which might need adjusting. This is only possible with the "Chase lock sync (time alignment by resampling) -> Active" function. If the timing of the master is inaccurate, audible pitch fluctuations may occur.

Note: This function should only be used if a common word clock is not available If a common word clock is available, synchronization variations will be created.

1. To synchronize via MIDI clock as slave, connect the MIDI OUT of the external device to the MIDI IN of the MIDI port selected in the synchronization setup.
2. Now select "MC Input active" in "Synchronization/MIDI clock input" and enter the tempo of the external sequencer. Samplitude is now operating as a slave. Unlike many other hardware sequencers, Samplitude does not deactivate the transport keys but “listens” to the incoming master commands.
3. Make sure that the external device is switched to "master synchronization" (many devices also call this "Internal sync mode" with "Sync Out" activated).
4. If you start the external device now, the VIP project in Samplitude starts, too.

If "Chase lock sync" is active, the fluctuations Samplitude has to compensate are displayed in the bottom right arranger window.

If your external sequencer is set to a different tempo than from Samplitude, this will attempt to adjust to the incoming tempo by changing the playback tempo relative to the initial BPM after starting the sequencer. However, this can take up to several bars and results in clearly audible pitch fluctuations. Make sure from the start that both synchronization units are set to the same tempo.

Application 3: MIDI time code (MTC) synchronization using a second software sequencer. Samplitude is master .

Samplitude can be synchronized with another software sequencer or hard disk recording system. MIDI time code synchronization is particularly useful for this.

The advantage of the synchronization method is that it is independent of a song's BPM entries. Rhythm and beat-dependent changes of tempo are therefore no longer a problem within a project, since communication now takes place via time values. Division into hours is possible according to the SMPTE code: minutes: seconds: frames (format: 00:00:00:00). Frames are the smallest SMPTE units. These originate from movie technology and state the number of images per second (e.g. 24 frames for film, PAL: 25 frames for video).

Note: If Samplitude should be synchronized with audio data and not film, then the frame rate can be set however you like. It's important that both of the systems to be synchronized feature the same sample rate.

Preparing for synchronization:

The second software sequencer can be running on the same or on a different machine as from Samplitude. If two sequencers are running parallel on one computer, make sure that the audio drivers being used permit simultaneous use of multiple sound card outputs. Alternatively, you can also use two different audio interfaces.

During synchronization via the MIDI interface, note that a MIDI port that is opened by the software cannot be used by other applications. To overcome this limitation, two MIDI interfaces that have been connected may be used. It's more elegant to use a freeware help program like "Hubis Loopback", "Marblesound Maple", or "MIDI Yoke". These programs install virtual, open MIDI ports on a PC that may be "connected" freely with the corresponding applications.

MIDI time code (MTC) synchronization using a second software sequencer. Samplitude is master:

1. Start both programs first.
2. Now connect the MIDI output of the MIDI port used by Samplitude with the MIDI input of the system to be synchronized, or patch this virtually using an auxiliary program.
3. Select the corresponding MIDI port in Samplitude's "MTC output" window and activate the MTC output.
4. On the slave system select the corresponding “slave” or “receive MTC” option. This setting is independent of the second program being used. Now, activate the used MIDI in-Port.
5. Ensure that the same frame rate is set for both systems.
6. Now start playback in Samplitude. The slave system will follow and be in sync with the master.

Important: Always make sure that the sync start position of the slave system is identical with the start position of Samplitude. If SMPTE sync start cannot be set for the other sequencer, you can also enter the starting time of the other system in Samplitude in "Synchronization -> SMPTE settings -> Offset".

Application 4: MIDI time code (MTC) synchronization using a second software sequencer. Samplitude is slave.

This method is generally the same as in point 3. However, the master/slave setting has to be reversed, i.e. MIDI OUT from the other sequencer to Samplitude's MIDI IN.

Backup Recording with Two Programs

In order to make sure that everything is recorded properly during important studio sessions, using parallel systems is strongly recommended. In the past the parallel system was normally a hardware device that recorded the stereo master or the complete multitrack setup. These days this type of backup can be recorded simply by using a second DAW.

Requirements for Backup Recording

- Two DAW systems e. g. Sequoia/Sequoia, Sequoia/Samplitude Pro X6 or Samplitude Pro X6/Samplitude Pro X6
- MIDI MMC Connection

You can let the backup system run uncontrolled. In this case you don't need an MMC connection. This technique runs the backup system the whole time and does not generate objects (takes). A timecode synchronization of the systems using MTC/SMPTE is not necessary as long as the digital recording and the word clock is identical in both systems.

Creating the Backup Recording

The MMC settings include the presets "Record Backup (Master)" and "Record Backup (Slave)". Set suitable presets on the master and slave systems.

Now set up an identical project for the master and slave, activate the necessary tracks and deactivate the display of the recording dialog at the end of the recording ("System Options" > "Record" > "Show confirmation dialog (OK, Delete) after record"). In the slave system, the option "Set cursor to record end position - next recording starts at this position" should also be activated. This will add each new take to the end of the project even when an insert punch is applied in the master.

Parellel Recording

For parallel recording set the MMC presets "Parallel Systems (Master)" and "Parallel Systems (Slave)".

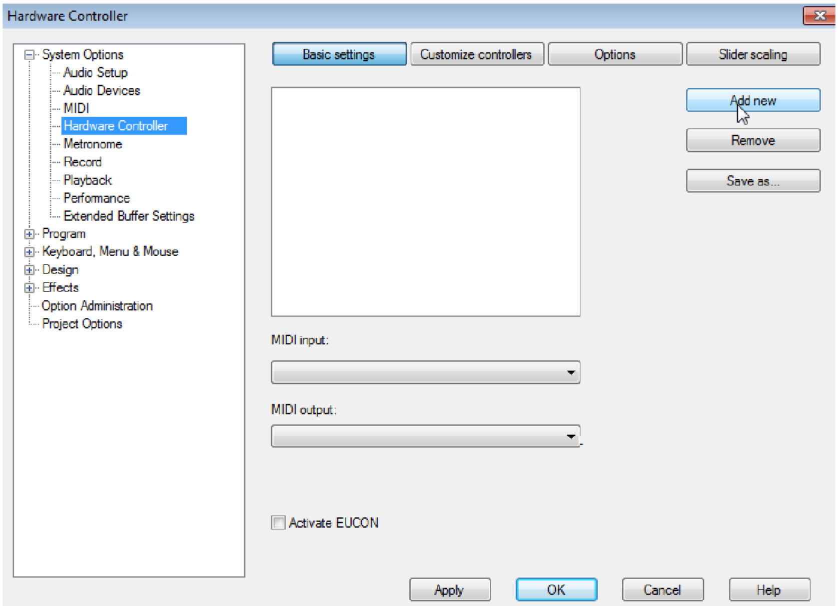
Now set up an identical project for the master and slave, activate the necessary tracks and deactivate the display of the recording dialog at the end of the recording ("System Options" > "Record" > "Show confirmation dialog (OK, Delete) after record").

In parallel recording the following information can be exchanged between the programs in addition to the transport functions: cursor position, mute, solo, input monitoring and record ready. You can adjust the settings to meet your own requirements.

In this way an almost identical project is created in the slave system whereby insert punch recordings are also added at the matching position. Only take names and markers are not taken into account.

Now if one of the systems fails during the session, you still have a complete project including the take structure and can continue working without having to add the takes again.

Hardware Controllers



Basic Settings

The following tasks are performed in the basic settings:

- Adding a hardware controller using a factory-preset (view page 494)
- Specifying MIDI ports for the hardware controller
- Determine the order and dependencies of the hardware controller: Hardware controllers can be operated independently or together as a unit.
- Adding a new hardware controller preset

Using EUCON Controllers

EUCON controllers are integrated through a network connection and contain an independent configuration environment.

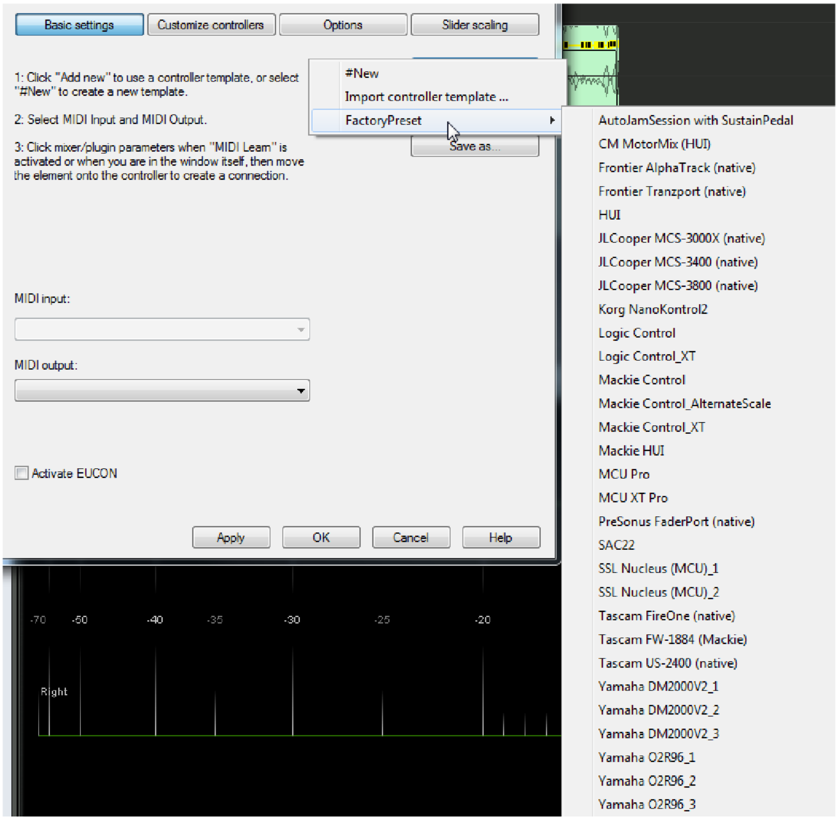
They are configured exclusively with the software included with the EUCON controller.

Activate EUCON: Activate this check box if you are working with an EUCON controller. The connection can be updated at any time using the corresponding button.

Adding a Hardware Controller using a Controller Preset

Tip: Many hardware controllers have a Mackie emulation mode. In this mode, most functions that are available in the Mackie Control are supported. If you are using this type of Mackie control emulation, simply select the preset "Mackie Control".

1. Click the "Add New" button in the controller settings.
   This opens the preset menu:



2. In the „Factory Preset“ list (view page 494) select your hardware controller or a compatible device.
   Result: The selected device appears in the list of hardware controllers.

Note: when you load a "Factory Preset", Samplitude saves this factory preset in a separate folder. Changes to this preset will then only be saved in this separate folder. This ensures that the factory presets are not changed and can be accessed again at any time.

Factory presets that have been loaded once appear additionally in the first level of the "Add New" menu. It is possible that adjustments have already been made to these presets.

3. Select the MIDI In and MIDI Out ports that your controller is connected to.
4. Click on the "Apply" button to confirm the changes.
5. If you want to edit the controller preset, open the "Customize Controller" view and edit the control elements as necessary (view page 479).
6. If you are using several controllers, repeat this procedure until all of the controllers have been edited.

Using Multiple Controllers, Extenders and Multibank Controllers

Note: When using several hardware controllers of the same type, add a separate preset to each controller. This ensures that the adjustments only have an effect on one of the controllers.

A maximum of four controllers can be positioned in the controller selection list. For example, if you select a Mackie Control and want to add an extension to it, add a "Mackie Control XT". This will be added automatically directly below the first one. In this case, the second controller is treated as the extension. Specify the correct MIDI ports for this additional controller.

1. Set up the controller as described above.
2. Use drag & drop to adjust the order of the controllers until they are actually in the right order.
   - Multiple controllers used in parallel are displayed on one level.
   - Expansion modules are displayed indented.
3. Activate or deactivate individual hardware controllers with a checkmark in front of the controller-entry. This can be useful for the following purposes:
   - Using Different Controllers for Various Applications / Environments
   - Functional Test of an Extender or Additional Controllers during Configuration
4. Click on the "Apply" button to confirm the changes.

Adding a New Empty Hardware Controller Preset

1. Select "#New" to create a new template.
2. Select the MIDI In and MIDI Out ports that your controller is connected to.
3. Click on the "Apply" button to confirm the changes.
4. Select the option "Save As..." and give the file a clear name.
5. Switch to the „Customize Controller“ (view page 468) view and map the control elements for your controller.

Removing a Controller from the List

Prerequisite: The list of used hardware controllers contains at least one entry.

Note: When a hardware controller is removed, the preset configuration remains. If you load the corresponding preset later, all of the settings for the hardware controller will be restored.

1. Simply click on the hardware controller that you want to remove.
2. Now click on the "Remove" button.
   Result: The selected hardware controller is removed from the list and is no longer used by Samplitude.

Customizing Controllers

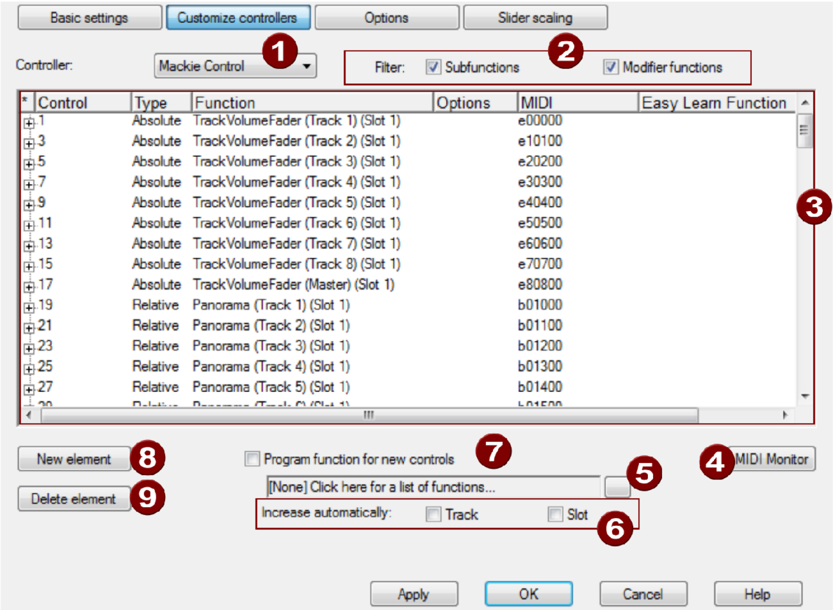
In the "Customize Controller" view you can make the following adjustments:

- Adjust already learned control elements
- Learn new control elements
- Apply or remove Easy Learn functions
- Remove already learned control elements

Note: when you load a "Factory Preset", Samplitude saves this factory preset in a separate folder. Changes to this preset will then only be saved in this separate folder. This ensures that the factory presets are not changed and can be accessed again at any time.

Factory presets that have been loaded once appear additionally in the first level of the "Add New" menu. It is possible that adjustments have already been made to these presets.

Controller Customization - Setup and Functioning



1 Controller
This is where you select the hardware controller that will be edited. When the "Customize Controller" view is opened, the controller that was last set in this view will be selected by default. Clicking on the dropdown menu opens a list of all currently available controllers.

2 Filters
Here you can specify whether sub-functions and modifier functions (view page 474) will also be displayed. Both of these options are activated by default.

Sub-functions: These are functions that belong to the control element but are accessed with a separate MIDI command.

- Push Function for Encoders
- Touch Function for Touch-Sensitive Faders

Modifier functions: Additional assignments to control elements. The modifier functions can be accessed by simultaneously pressing the assigned modification button and the corresponding control element key.

3 **List of Mapped Control Elements:** This list contains all of the control elements that have already been mapped.
The list is made up of several columns where the entries can be sorted by clicking on the column headers:

*: If the control element has sub-functions or modifier functions, they will be displayed in this column.

Note: If the list is sorted according to this column (*), the assigned or new control element will be displayed at top of the list when a MIDI command is received.

Element: Automatically numbered list of the control elements. Sub-functions and modifier functions are displayed here in color and with a corresponding short description:

Sub-functions: Touch/Push - Displayed in green

Modifier functions: Shift / Control / Option (also in combination) – Displayed in red.

Type: Displays the control element type (view page 471) that is assigned to an entry.

Function: Displays the function that is assigned to the control element.
Right-clicking opens the "Edit Function" dialog. The program functions that should be assigned to control element are set in this dialog.

Options: Displays whether various options for this control element are active.
Right-clicking opens the context menu where the available options can be selected.

MIDI: If the function is already assigned to a control element, the MIDI message that opens the assigned function is displayed here.
Right-clicking opens the "Select MIDI Data" dialog.

Easy Learn Function: If a control element is assigned an additional function with Easy Learn, the function will be displayed here.
Right-clicking opens the context menu where the function can be permanently assigned or deleted.

4 **MIDI Monitor** (view page 475)
This button opens the MIDI-input monitor.

When the Shift key is pressed during opening, the HWC MIDI-output monitor is also opened.

5 **Dropdown Menu Function**
The program functions that should automatically be assigned to the next newly added control element are set in this dialog. The "Program Function for New Elements" checkbox must be activated for this.

6 Raise Automatically (Track/Slot)
The track or slot assignment of new control elements to be mapped is automatically raised when the previous control element is mapped.

This option is suitable for mapping control elements with which the same function should be controlled on different tracks/slots.

7 Program Function for New Elements
When this checkbox is activated, the next new added element will be assigned to the set underlying function.

8 New Element
This button sets a new entry in the list. The entry appears at the very bottom of the list.

9 Delete Element
This removes the selected control element from the list.

Types of Control Element

Value Input

- **Absolute:** Control elements with absolute values.
  This type should be selected when using a fader or knob.
- **Relative:** Control elements with relative values.
  This type should be selected when using an endless encoder, the jog/shuttle or a similar control element.

Note: Control elements for the values are already assigned tomodifier functions (view page 474).

Keys

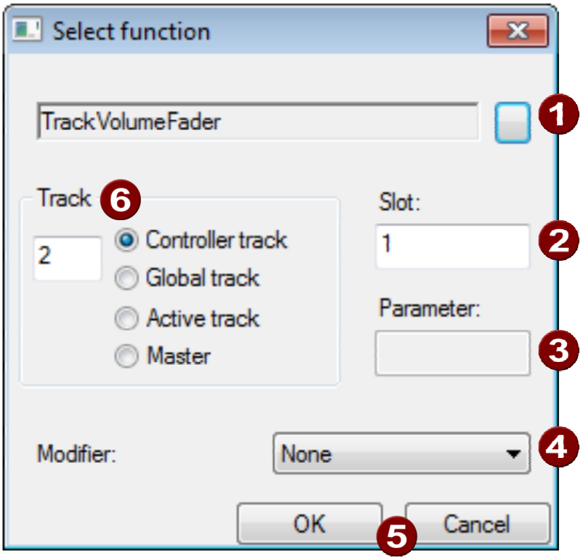
- **Button:** Simple keys
  Select this type of key when the keys on the hardware controller send a Note On command when pressed and a Note Off command when released.
- **Push Button:** Keys without a MIDI-command when released
  Select this type of key when the keys on the hardware controller ONLY send a Note On command when pressed.
- **State Button:** Key with a rest function
  Select this type of key when the keys on the hardware controller send a Note On command when pressed and a Note Off command when pressed a second time.

Other

- **LED:** Display function on the hardware controller
  LEDs or status lights are often integrated into other control elements or assigned to them. For this reason the "LED" status is most often changed using the same

MIDI command that sends the corresponding superordinate control element to Samplitude.
If a status display is not assigned to any control element, or is changed with a different MIDI command, please refer to the hardware controller manual to find out the correct MIDI command for the LED.

Select Dialog Function



1. **Function selection:** This is where the function assigned to the control element is displayed.

   The function that should be assigned to the control element is selected in the dropdown menu.

2. For certain functions there are several slots per track, e.g. for AUX sends, the equalizer (one slot per EQ band respectively) and the insert effects.

   Slot: Determines the number of the slot that the assigned control element affects. For functions that are only present once per track, this parameter does not cause any change.

3. **Parameter:** Additional dimensions for the selection of specific track parameters. If an effect has several parameters, the actual parameters are numerically ordered in this field.

   For example, the following scheme would be used for the track equalizer:
   - **Slot:** Selection of frequency band (= 4 Slots)
   - **Parameter:** Selection of parameter in the respective frequency band (Gain/Frequency/Q)

4 **Modifier:** The control element is only active when the set modifier key or the set combination is pressed.

The function opened by the modifier is called the modifier function (view page 474).

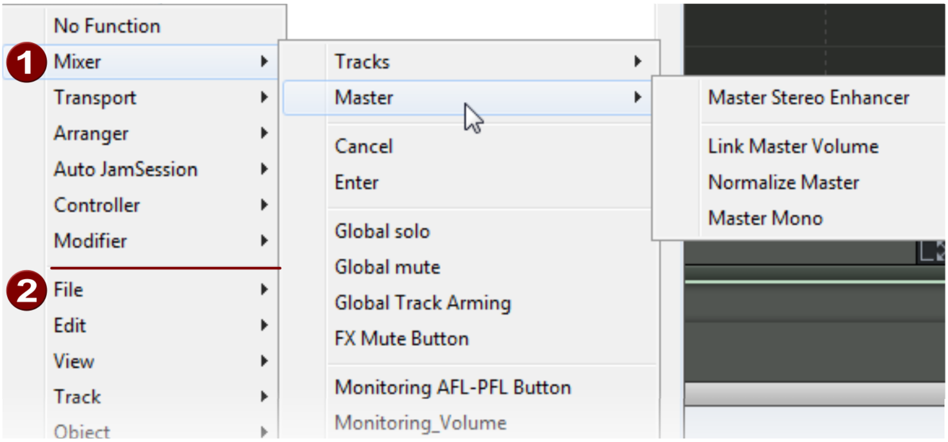
5 **OK:** The dialog is closed and the changes are applied.

Close: The dialog is closed without applying the changes.

6 **Track:** Track-based functions are assigned to the track through this parameter.

- **Controller Track:** The track allocation of the assigned function is controlled by the setting „Synchronize with“ (view page 485) in the „Options“.
- **Global Track:** The assigned function always affects the entered track.
- **Active Track:** The assigned function always affects the active track.
- **Master:** The assigned function always affects the master track.

Function Selection



1 The upper area contains the functions that are typically controlled with hardware controllers.

2 The lower area displays the complete Samplitude menu structure. You can use this to assign each menu command to a controller key.

Note: Please note that when working with menu commands, the key LEDs of the controllers remain inactive.

Modifier Functions

Modifier functions are multiple allocations of control elements that are only active while the modifier key is being pressed. These hardware controller modifier keys first have to be mapped (view page 482) in order to function. Any key on the hardware controller can be used as a modifier key.

The following modifier keys are available.

- Shift (alternatively locked)
- Option
- Control

Note: „Shift Lock“ has the same effect as „Shift“ but the corresponding key remains "snapped" when the key is released. Pressing the "Shift" key again turns the lock off. "Shift Lock" is particularly useful when combined with other modifier keys or when the status of the Shift key is displayed by the hardware controller.

As soon as one or more modifier keys is pressed, the respective additional allocation of the correspondingly mapped control element becomes active. After releasing the modifier key, the standard function becomes active again.

Combinations of Modifier Keys

The modifier keys can also be used in combination which increases the scope of remotely controlled commands even further.

The following combinations are available:

- Shift (Lock) + Option
- Option + Control
- Shift (Lock) + Control
- Shift (Lock) + Option + Control

Preset Modifier Functions

For Control Elements for Value Entry (view page 471) there are preset modifier functions:

- Shift (Lock) + FaderTouch (without moving the fader): The allocated parameter is reset to the default value (e.g. track volume: 0.0 dB).
- Shift (Lock) + Moving endless knob: A finer value gradation during a change to the value.

MIDI Monitor

The MIDI monitor shows either MIDI input data or MIDI ouput data.

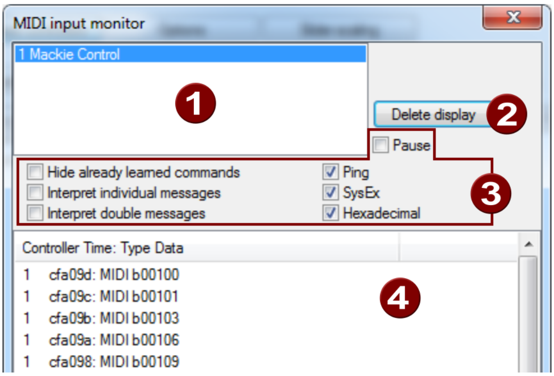
- MIDI Input Monitor: Shows all MIDI data being sent to Samplitude from the hardware controller.
  This dialog is modal, i. e. no other dialogs can be used when the MIDI monitor is open.
- HWC MIDI output monitor shows all MIDI data being sent from Samplitude to the selected the hardware controller.
  This dialog is not modal. This means that Samplitude is fully operational while the MIDI monitor is open.

The MIDI data that is displayed depends on the settings.

Open

- **MIDI Input Monitor:** Activate the „MIDI Monitor" button.
- **HWC MIDI Output Monitor:** Activate the „MIDI Monitor" button while holding down the Shift key.

Setup and Functioning



1 **Selection of Hardware Controller for MIDI Monitoring:** Clicking on the entries activates or deactivates the hardware controller for MIDI monitoring.

2 **Delete Display:** This deletes the values in the display field (4).

3 Options

- **Hide already learned commands:** MIDI commands that have already been mapped in Samplitude won't be shown in the list.
- **Interpret individual messages:** The MIDI commands are interpreted as 7 bit values.
- **Interpret double messages:** 2 MIDI commands are interpreted as 14 bit values.
- **Ping:** If this option is activated, the display of the MIDI Ping command entered under "Options" (view page 484) is disabled.
- **SysX:** If this option is activated, SysEx data is displayed.
- **Hexadecimal:** The MIDI commands are displayed in hexadecimal format. If this option is deactivated, the MIDI commands are displayed in decimal format.
- **Pause:** If this option is activated, no new data is displayed.

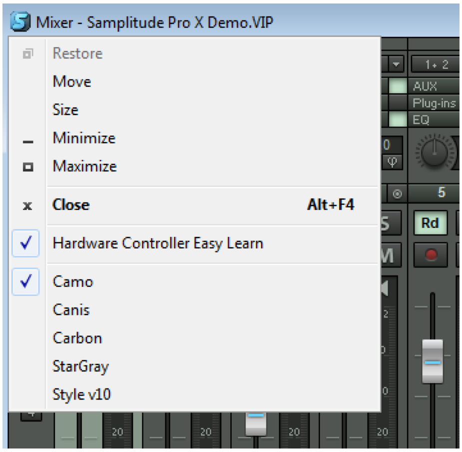
4 **Display range for MIDI data:** The MIDI data that is sent and received from the hardware controller is displayed here.

New MIDI data is displayed at the top of the list.

Controller Learning with Easy Learn (Plug-in and Mixer Elements)

The respective active mode can be adjusted for individual settings. The mixer system menu features the option "Hardware Controller Easy Learn" for this. Right-click on the title bar in the mixer or click on the icon to open this function.

Note: For assignment using Easy learn, only control elements that have been learned in the hardware controller dialog can be used.
Assignment using Easy Learn changes the active mode until the option "Restore Modes" is selected in the „Options“ (view page 484) menu.



1. Click on the command „Hardware Controller Easy Learn“.
2. Activate a mixer element with the mouse.
3. Now move the desired control element on the controller.
   Result: The controller control element now controls the newly assigned mixer element.

The plug-in menus also include the option "Hardware Controller Easy Learn". The parameters of a plug-in can also be mapped to a controller in the same way.

The changes remain available until

- the option „Restore Modes“ is selected in the hardware controller dialog or
- a change is made to the hardware controller dialog settings.

Here are the instructions for mapping a hardware controller:

1. Select the ports you used to connect the controller under MIDI input and MIDI output.
2. If there is already a template for your controller in the "Factory Presets", you can select this and your controller will be instantly ready for use.
   In this case you can skip the following steps.
3. Create a new controller entry by clicking on „Basic settings“ then „Add New" and selecting the option "#New". If a prompt for activation of the MIDI SysEx input then appears, answer this with "Yes".
4. If you haven't yet set up any connection between your controller and the PC, do that now. Most controllers are either connected by USB or MIDI.
5. In the "Options" tab, select "Generic MIDI" as the protocol and enter the number of control elements on your controller as the number of tracks (e. g. if your controller has 8 faders, enter "8").

> **Note**: If you are not sure, leave this value set to "0". You can also leave the other values in their default settings.

6. Now switch to the "Customize Controllers" tab.
7. Here you can first assign all of the control elements that you want to use. The following process is the same for every control element:
   - Open the mixer or plug-in you want to control and move the desired parameters with the mouse.
   - Now move the desired control element on the hardware controller. After 1 or 2 seconds it will appear in the Element List. The assigned parameter now appears in the "Easy Learn function" column of the control element.
   - Then check that the type has been correctly set and correct the setting if necessary (e. g. non-motorized faders and knobs should be set to the "Absolute" type).

Notes and tips:

- As soon as you have opened the hardware controller settings, you are automatically in the Easy Learn Mode, i.e. every mouse movement of a mixer or plug-in parameter causes the assignment of this mixer or plug-in parameter to the last touched control element of your hardware controller.
- When you are finished assigning the controls be sure to shut "Hardware Controller Easy Learn" off to avoid making any unwanted changes to the settings.
- If it is not possible to assign a control element, first check in the "MIDI Monitor" ("Customize Controllers" tab) whether MIDI data is being received at all when moving the element. If this is not the case, your controller is either wrongly

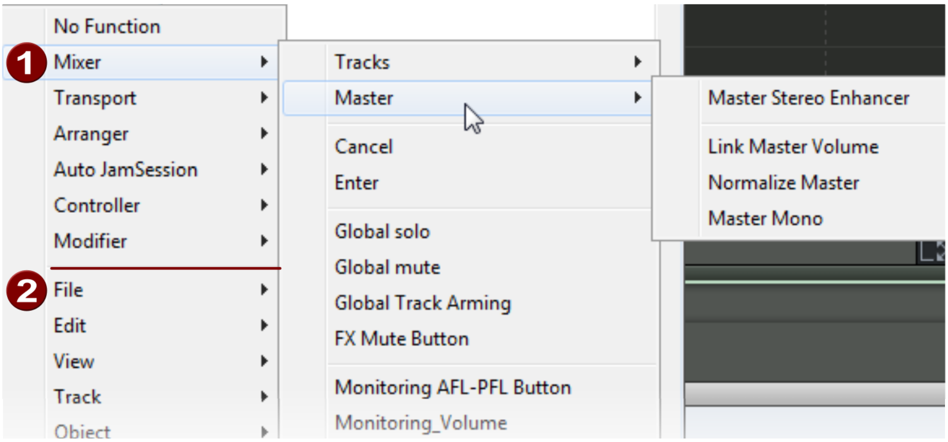
configured (wrong MIDI connection or incorrect internal mode of the controller) or the relevant control element is not sending data (other elements work). If the latter is the case, unfortunately you cannot use the element.

- In addition to the normally assigned "function", every control element can also have the "Easy Learn Function".
- The normal function is permanently assigned to the element whereas the "Easy-Learn Function" is a temporary function that has priority over the normal function if present.
- - You can also apply this after mapping an "Easy Learn Function". To do this, click with the right mouse button on the relevant element in the "Easy Learn Function" column. The "Easy Learn Function" is now transferred to the normal function.

Adjusting the assignment of a control element

Assigning a new function

1. Right-click on the desired entry in the "Function" column.
   The "Select Function" dialog will open.



2. Open the dropdown menu and select the desired function.
3. Click "OK" to close the dialog.
4. Click on the "Apply" button to confirm the changes.
5. Test the assignment by moving the control element on the hardware controller.

Mapping control elements

Mapping mixer or plug-in control elements

All of the control elements on the mixer or plug-ins that should be controlled by the hardware controller can be mapped using the Easy Learn function.

Requirement: The „Hardware Controller Setup“ dialog is opened in the "Adjust controller" view.

To do this, proceed as follows:

1. Select the mixer element that will be assigned to the control element.
2. Now move the control element on the hardware controller.
   Result: A new entry will be added to the list of control elements. The MIDI command that will be sent from the hardware controller is displayed in the "MIDI" column.

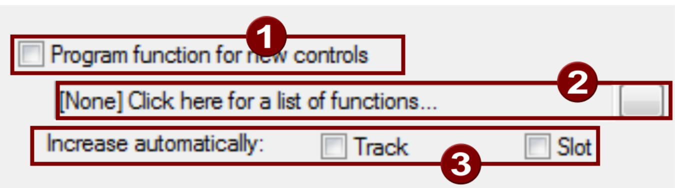
> **Note:** If the control element on the hardware controller is touch-sensitive, a fader touch message will be sent when it is touched and when it is released. In this case 2 new elements are added.

3. Repeat this procedure for all of the control elements that should be mapped.
4. Select all of the entries in the list.
5. Move the mouse pointer over the entries in the "Easy Learn Function" column.
6. Right-click to open the context menu and select the option "Apply Easy Learn Function".
   The entries will be assigned automatically.
7. Click on the "Apply" button to confirm the changes.
8. When additional fader touch messages were added (button): Assign subfunction (view page 482).

Adding the same control element to multiple tracks/slots

Hardware controllers capable of simultaneously controlling multiple tracks/slots can be mapped quickly and easily.

To do this, proceed as follows:



1. Activate the option „Hardware Controller Easy Learn“ in the Mixer's menu.
2. Activate the option „Program Functions for New Controls“ ❶.
3. Activate the option „Increase Automatically: Track/Slot“ ❸.
4. Assign the desired function:

    1. Click on the field for the function selection❷.
       The "Select Function" dialog will open.
    2. Open the dropdown menu and select the desired function.
    3. If necessary, activate the desired modifier key in the dialog.
       The mapped function is then only activated when the set modifier key is also pressed.
       The modifier key must also be mapped (view page 482).
    4. Click „OK“ to close the dialog.

    Note: Hardware controllers with touch-sensitive faders send a message whenever they are moved and released (FaderTouch). In this case 2 entries are automatically added, but the entry for the fader touch message is not assigned.

5. Move all of control elements of this type on the hardware controller.
   When the control elements are moved, a new entry is added and the track/slot number is increased by 1.
6. If additional fader touch entries are generated, assign these as subfunctions (view page 482).
7. Deactivate the option „Hardware Controller Easy Learn“ in the Mixer's menu.
8. Click on the "Apply" button to confirm the changes.

Mapping individual control elements

Control elements can be mapped individually, e.g. for special program functions. The modifier keys are also mapped individually.

To do this, proceed as follows:

1. Manipulate the desired control element on the hardware controller.
   A new entry will be added to the list of control elements. The MIDI command that will be sent from the hardware controller is displayed in the "MIDI" column.

   Note: Hardware controllers with touch-sensitive faders send a message whenever they are moved and released (FaderTouch). In this case 2 new elements are added.

2. Assign the desired function:

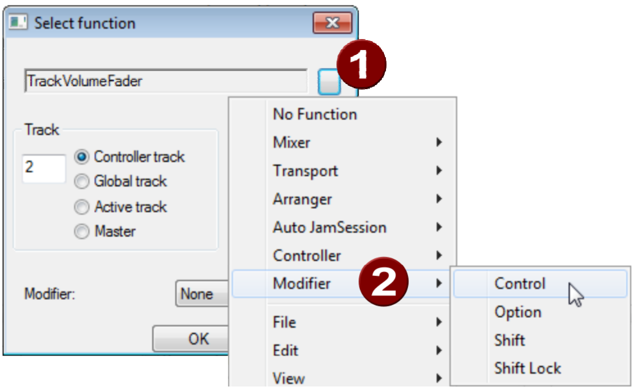
    1. Right-click on the list entry in the "Function" column.
       The "Select Function" dialog will open.

2. Open the dropdown menu and select the desired function.
3. If necessary, activate the desired modifier key in the dialog.
   The mapped function is then only activated when the set modifier key is also pressed.
   The modifier key must also be mapped (view page 482).
4. Click "OK" to close the dialog.

3. Repeat this process for all of the control elements that should be mapped.

Mapping modifier keys

Mapping modifier keys is exactly the same as mapping individual control elements (view page 481) of a hardware controller.

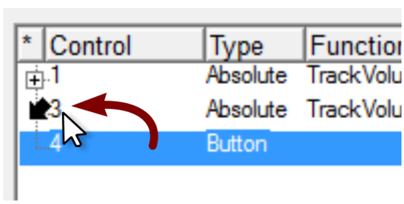


The modifier keys can be found in the "Select Function" dialog in the function selection menu 1 under "Modifier" 2.

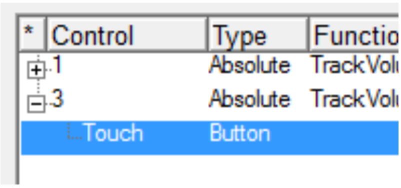
Assigning subfunctions

> **Tip:** Assigning elements can be simplified if they are displayed in the order in which they were detected.
> Select the "Elements" column to sort the detected control elements in this order.

1. Manipulate the control element on the hardware controller without moving it.
The corresponding entry will be selected in the list.



2. Now drag & drop the entry onto the corresponding entry for the movement, e. g. „TrackVolumeFader (Global Track 1) (Slot 1)“.
   A popup menu will open.
3. Here you can specify the subfunction:
   - When dealing with a touch-sensitive control element such as a touch-sensitive fader:
     Select the entry „Insert as touch element“,
   - When dealing with a control element with additional key function such as an encoder that can be additionally pressed:
     Select the entry „Insert as push element“,



Assigned touch function of a touch-sensitive fader

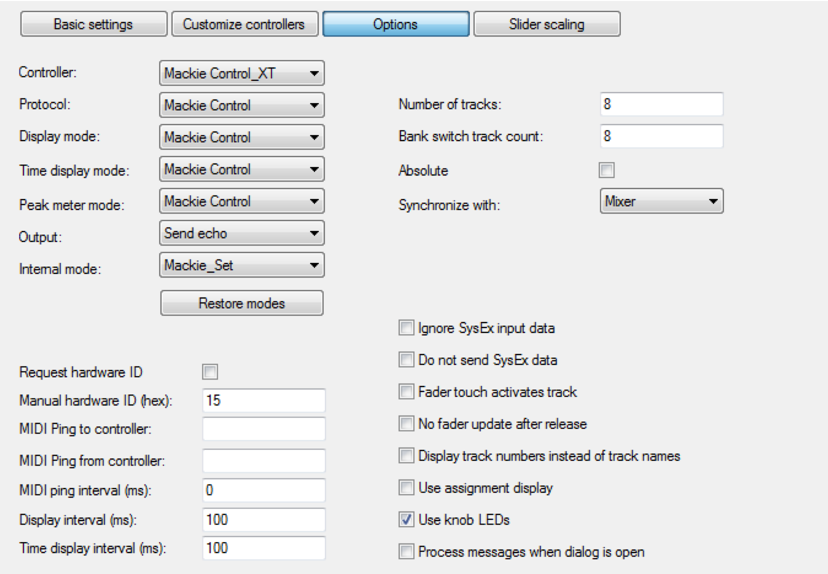
Correcting incorrectly recognized control elements

In rare cases some control elements might be improperly recognized during mapping, e.g. as "relative" instead of "absolute". This can be caused by various factors, e.g. large value jumps when moving the fader.

You can correct these incorrectly recognized control elements as follows:

1. Select the incorrectly recognized control elements:
   - with <Shift> for multiple control elements positioned one after the other
   - with <Ctrl> for multiple control elements
2. Position the mouse pointer over one of the entries in the "Type" column.
3. Right-click to open the context menu.
4. Select the appropriate type for the control element (view page 471).
5. Click on the "Apply" button to confirm the changes.

Options



Received MIDI messages are interpreted differently depending the the type of controller being used. In the "Options" view you can adjust the settings for the controller behavior to fit your personal workflow.

General Options

Controller

The hardware controller whose options will be edited is selected here. The list contains all hardware controllers that have been set in the "Basic Settings" view.

Protocol

The protocol determines how Samplitude communicates with the hardware controller. This determines the structure as well as the number and the order of the MIDI messages that make up a command.

The following protocols are available:

- Generic MIDI
- Mackie Control
- HUI
- JLCooper
- Tascam 2400
- FaderPort

Samplitude options

Number of tracks

Number of tracks: Specifies the number of tracks in a bank available to the controller. This is usually eight tracks. Mackie Control and HUI protocols can only communicate with eight channels per bank. However, some controllers have a native mode which lets you learn several channels (64 max.). For single track controllers, the value is "1".

Number of tracks for bank switch

A project in Samplitude can include more tracks than are available to the controller. In this case the controller can only display a section of the available tracks. To change this section you can perform a channel switch or bank switch. A channel switch changes the section by one track while a bank switch changes the section by several tracks. You can set different step sizes for each bank depending on the number of tracks on your controller and your preferred working method.

Synchronize with

The following options deal with the synchronization between the controller area, mixer area, and the active track. The controller area corresponds to the section of available tracks in the project that can be operated with the controller. The mixer area is the section of available tracks in the project that can be controlled with the Samplitude mixer.

Synchronize with Active Track: If the active track is changed, the controller area will be adjusted so that the active track is within the controller area.

Synchronize with Mixer: If a change is made to the controller area, the beginning of the mixer area will be adjusted according to the beginning of the controller area.

Note: Make sure that the synchronization of the view in Samplitude with the hardware controller is dependent on the control element settings (Track) (view page 472).

Fader touch activates track

This option activates the track that belongs to the fader in the application when a touch-sensitive fader is affected.

Use assignment display

Some controllers have an additional display field for showing the current operational mode (track, EQ, AUX, etc.). If a controller does not have this kind of display, you can deactivate this option to prevent the control commands from incorrectly influencing other controller elements.

Process messages when dialog is open

Normally, when hardware controller dialogs are open, MIDI messages are no longer processed so that unwanted changes are avoided. When a controller is being customized or programmed it is sometimes necessary to send back messages to the controller. In this case, you can activate incoming messages while the dialog is open. Make sure to deactivate this feature when you are finished customizing the controller.

Options for controlling the hardware controller

Display mode

Many hardware controllers have LCD displays which differ in size and the number of lines and which need to be controlled differently. For hardware controllers without a display you can simply set the display control to "Off".

The following display modes are available:

- Off
- JLCooper
- Mackie Control
- HUI

Display interval (ms)

Display control occurs in certain time intervals and not continuously. This is meant to limit the amount of display information. A short interval accelerates the display on the controller but increases the amount of MIDI data that must be transferred.

Show track numbers instead of track names

If this option is active, track numbers will be shown instead of track names in the controller display.

Time display mode

Some hardware controllers have a special display for the current time position (Locator). The time display mode can be set for the following types of display:

- Off
- JLCooper
- Mackie Control
- HUI

Time display interval (ms)

Locator display control also takes place in time intervals. If the time position changes within an interval, only the last current value will be sent to the controller. If you shorten the interval, the locator display will react faster and more directly.

Peak Meter Mode

Some hardware controllers have a special peak meter display. The peak meter mode sets the controls for this display.
The following peak meter modes are available:

- Off
- Mackie Control
- HUI
- Tascam 2400

Output

- **Deactivate:** Hardware controller data is received and processed, Samplitude doesn't send any data back to the controller.
- **Normal Send:** The same as „Deactivate“, but changes in Samplitude are sent to the hardware controller.
- **Echo Send:** The same as "Normal Send" except that the received hardware controller data is immediately sent back and end at the controller, e.g. data for status LEDs.
- **Echo Send (without fader touch):** The same as "Echo Send", but no fader values are sent while the faders are being moved. The fader value is only sent back to the hardware controller when the fader is released.

Note: Some controllers (e. g. „Logic Control“) have problems receiving echo messages (termination). In this case select the setting "Normal Send".

Do not send SysEx data

Samplitude sends different messages via SysEx. Exactly what is sent depends on the selected display mode, locater mode etc.

If SysEx messages start to cause problems, you can prevent these kinds of messages from being sent to the controller.

No fader update after release (Motorized Fader)

When you move touch-sensitive faders, the fader messages to the controller are normally ignored by the controller. After release the hardware controller expects a position message as termination.

If this message is not sent, the fader will slide back to the position it was in before it was moved. Some controllers do not need this new position message. When they receive the new position they first jump to the old position and then to the new one.

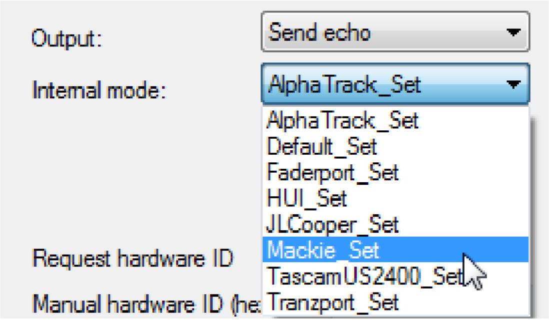
If the fader update after release of the fader should not be sent, you can deactivate this sending of the new position here.

Control LED knobs

Some controllers have LED displays that are assigned to the knobs. These can be used to visually display the current parameters of the knob. If you are using a controller with no knob LEDs, you can deactivate this option.

Options for the interpretation of hardware controller demands

Hardware Controller - Internal Mode ("Options" tab)



Note: The parameter „Internal Mode“ is only available for hardware controllers that have been added to the "Basic Settings" view on the first level and therefore act as the "Main Controllers".

The "Internal Mode" determines how Samplitude reacts to incoming remote commands from hardware controllers and enables adaptation to conceptually different hardware controllers.

Depending on the mode, different information is displayed on the controller and Samplitude interprets the controller messages correspondingly. The effect is that a particular physical control element controls different parameters in Samplitude, depending on the internal operating mode.

These data structures can be manipulated using the function „Hardware Controller Easy Learn“ in the mixer menu and the menus of the VST plug-ins (view page 476). In this case, the active operating mode can be changed so that it is able to function differently than its preset. The changed data structures are saved accordingly and are immediately available the next time the program is launched. This ensures that you are able to continue working with the most recent settings.

Note: If you are using a hardware controller that is not listed in the hardware control setup, select one from the list that is most similar in concept and design.

The following internal modes (Mapping Modes) are available:

Frontier AlphaTrack_Set

Track

Press first time: Panorama mode

- Panorama control
- Navigate to markers
- Move selected track

Press second time: Active control mode

- Changes the value of the selected element in the mixer with the fader.

AUX

- Select other track
- Select other slot
- Change AUX send of the selected track/selected slot

EQ

Controlling the input volume, frequency, and quality bands of the selected EQs. The individual bands are selected with "Mixer track up/down".

Plug-in

- Select track
- Select plug-in slot in the track with "Mixer track up/down" key

Now you can control 3 parameters of the corresponding plug-ins with the vpots. The display page is toggled with the keys "Mixer track up/down".

Default Set

Presonus FaderPort_Set

Active control

The currently active mixer element (recognizable by red highlighting) is controlled by the first fader.

Mackie HUI_Set

Track

Track mode

Active control

The currently active mixer element (recognizable by red highlighting) is controlled by the first fader.

EQ (Phat Channel EQ mode)

Controls the EQ for the selected track via the knob controllers.

AUX (Phat Channel AUX mode)

Controls all AUX sends in the selected channel.

AUX [slots] (1 - 8) (Track AUX mode)

Controls AUX sends for the different tracks in the selected slot.

JLCooper_Set (MCS)

Track

Press first time: Track mode

Press second time: Active control mode

- Each activated mixer element (recognizable by the red marking) is controlled by the first fader

AUX

Controls all AUX sends in the selected channel.

EQ

Controls the EQ of the selected track via the knobs.

Mackie Multitrack

Track

Press first time: Track mode

Press second time: Active control mode

- Each activated mixer element (recognizable by the red marking) is controlled by the first fader

PAN

- Pan mode (like track mode, except for the display)

EQ

Press first time: EQ Phat Channel Bandwise mode

- Gain, frequency, and quality accessible via the knobs
- Use the bank switch keys to access the other bands

Note: The bank switch functions have to be learned, which is usually the case

Press second time: EQ Phat Channel Typewise mode

- Gain, frequency, and quality accessible via the knobs (differently grouped than in Bandwise mode)
- Using the bank switch keys, each of the other EQ functions can be accessed

AUX

Press first time: AUX track mode

- AUX 1 to 6 of the active track using the knobs

Press second time: AUX slot mode

- Set AUX for each track
- Select number of AUX tracks to be controlled using the Bank Switch keys

Plug-in

- The VST plug-in slots are listed for the active track (flip through via bank switch keys)
- Selection of the VST plug-in to be adjusted by pressing the corresponding knob
- Thereafter the parameters for the selected plug-ins are shown
- Changing the parameter using the knob
- Page through the parameters

Frontier Tranzport_Set

Functions: For transport only

Request hardware ID

Activate or deactivate the automatic hardware ID request. Deactivate this function if you aren't using the "Mackie Control" display mode. This helps to avoid incorrect interpretation of the request message.

Manual Hardware ID

For display in "Mackie Control" display mode, an additional parameter is necessary: The hardware ID. This can be automatically requested with new firmware versions or manually entered

MIDI Ping to controller

Most controllers expect a periodic incoming MIDI message (ping) to ensure that there is communication with the application. Different messages can also change the operating modes of a controller. Here you can enter the message that should be sent to the controller.

MIDI Ping from controller

In response to ping messages, controllers usually send a ping message back. To ensure that these are not falsely mapped as MIDI commands you can enter these explicitly so that they are ignored by the application.

MIDI Ping interval (ms)

Here you can specify the time period between the two ping messages in milliseconds.

Ignore SysEx input data

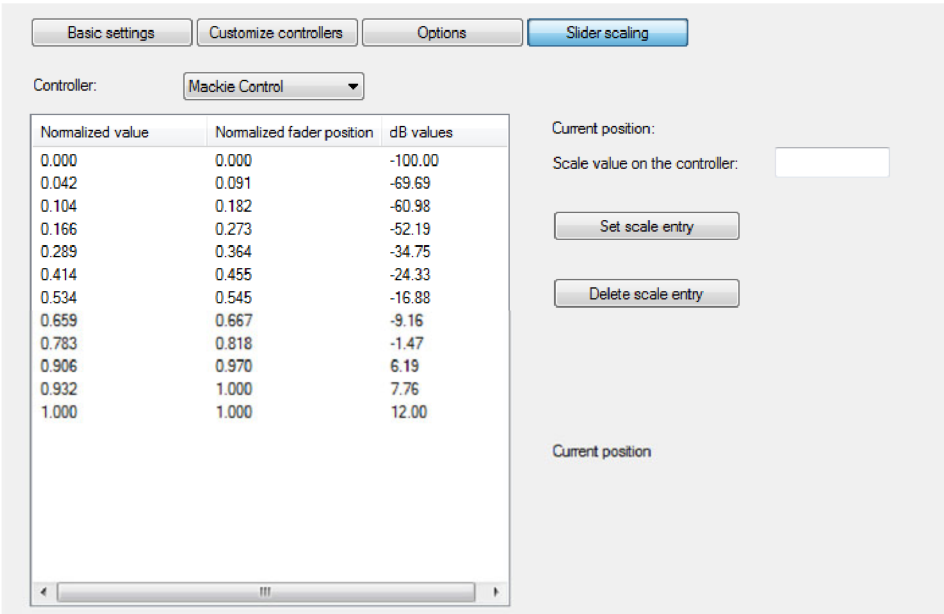
SysEx data from the hardware controller will not be processed by the application.

Fader Scaling

Fader scaling offers you the option of calibrating the scale on the hardware controller to the volume fader values in Samplitude.

Control with a hardware controller can take place in a range from +12 dB to -100dB. Normally at +12 dB the fader position is at the very top and at -100 dB at the very bottom.

Note: Calibration of the fader scale only makes sense when the fader scale on the controller actually includes numerical values.



Add scale values

1. In the „Controller“ section select the hardware controller whose printed scale should be calibrated in Samplitude.
2. Delete all previous entries. Click on the entry in the list and then press "Delete scale entry".
3. Enter the value "6" into the "Scale entries on controller" and move the fader all the way up (to the scale value +6dB). Next, click "Set scale entry" to set the new entry.
4. Move the fader all the way to the bottom and enter the lowest scale value (at least „-100“) in the field „Scale entry on controller“. Click "Set scale entry" again.
5. Enter further scale positions in the same way (e. g. -12, -24, -48...).

Test the mapped scale

1. Click "OK" to close the dialog.
2. Compare the entered fader position of the hardware controller with the fader position in the program mixer.

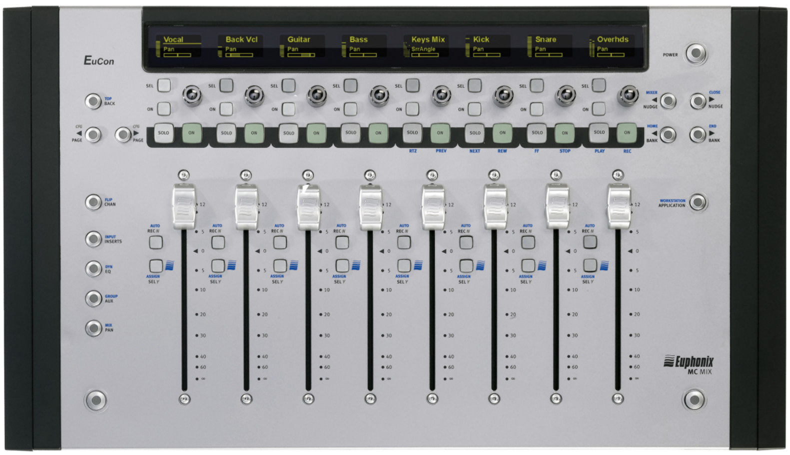
Changing entered scale values

1. Select the incorrectly mapped value
2. Click on the button "Delete scale entry"
3. Add the correct scale value.

Hardware Controller Presets

The following is a list of the most important presets for hardware controllers including their integration and configuration.

Artist Mix



1. Close your controller (according to manufacturer's instructions) and install the required driver software.
2. Go to System Options/Hardware Controller and select "Activate EUCON".
3. Close the dialog. Your controller is now ready to use.

Note

The modes INPUT, DYN, GROUP and MIX are not available in the software.

Track range

Control element	Function	Shift function
SEL 1~8	AUX: on/off, EQ: Switch between Freq and Q	
ON 1~8		
KNOB 1~8	Depending on mode (PAN, AUX, EQ, INSERTS)	
SOLO 1~8	Track solo	
ON 1~8	Track is on (inverted mute)	
RTZ		Cursor at project start
PREV		Previous marker
NEXT		Next marker
REW		Fast rewind
FF		Fast forward
STOP		Stop
PLAY		Playback
REC		Recording
FADER 1~8	Track volume	
AUTO REC N	Track record	Track automation
ASSIGN SEL Y	Track selection	Arrange tracks using EuControl

Navigation

Control element	Function	Shift function
TOP BACK	Return from a subpage	Return to upper most page
< PAGE	Parameter page to the left e.g. in channel editing mode, AUX or Inserts	
PAGE >	Parameter page to the right e.g. in channel editing mode, AUX or Inserts	

Control element	Function	Shift function
FLIP CHAN	Channel editing mode (Phat-Channel)	Flip mode KNOBS and FADERS swap functions
INPUT INSERTS	Plug-in Mode	
DYN EQ	EQ mode	
GROUP AUX	AUX mode	
MIX PAN	Pan mode	
SHIFT	Shift key	
MIXER < NUDGE	Controller track 1 to the left	Mixer
CLOSE NUDGE >	Controller track 1 to the right	Close application
HOME < BANK	Controller track 8 to the left	View goes to track 1
END BANK >	Controller track 8 to the right	View goes to last track
WORKSTATION APPLICATION	Switch the current Windows application	Workstation switch

Artist Control

1. Close your controller (according to manufacturer's instructions) and install the required driver software.
2. Depending on the program version, copy the files "Sam.xml" or "Sequoia.xml" from the program folder „/Customize/EUCON“ to „Programs\Euphonics\EuCon\SQRL“ or to the location of your user template.
3. Go to System Options/Hardware Controller and select "Activate EUCON".
4. Close the dialog. Your controller is now ready to use.

Note

The modes INPUT, DYN, GROUP and MIX are not available in the software.

Track range

Control element	Function	Shift function
SOLO 1~4	Track solo	
ON 1~4	Track is on (inverted mute)	
FADER 1~4	Track volume	
AUTO REC N	Track record	Track automation
ASSIGN SEL Y	Track selection	Arrange tracks using EuControl
HOME SOFT KEYS	SOFT KEYS upwards	SOFT KEYS to first page
END SOFT KEYS	SOFT KEYS downwards	SOFT KEYS to last page

Display area

Control element	Function	Shift function
KNOB 1~8	Function independent of view	
SOFT KEYS 1~12	Function independent of view	

Navigation

Control element	Function	Shift function
MONITOR CONTROL ROOM	Monitor volume	Solo-Bus volume
TOP BACK	Return from a subpage	Return to upper most page
WORKSTATION APPLICATION	Switch the current Windows application	Workstation switch
< PAGE	Parameter page to the left e.g. in channel editing mode, AUX or Inserts	

Control element	Function	Shift function
PAGE >	Parameter page to the right e.g. in channel editing mode, AUX or Inserts	
MIXER < NUDGE	Controller track 1 to the left	Mixer
CLOSE NUDGE >	Controller track 1 to the right	Close application
HOME < BANK	Controller track 8 to the left	Display switches to Track 1
END BANK >	Controller track 8 to the right	Display switches to last track
REC	Recording	
PLAY	Playback	
STOP	Stop	
FF	Fast forward	
REW	Fast rewind	
SHUTTLE JOG	Positioning	Scrubbing
VERTICAL ZOOM HORIZONTAL ZOOM		
SHIFT	Shift key	

Artist Transport

1. Close your controller (according to manufacturer's instructions) and install the required driver software.
2. If the files "Sam.xml" or "Sequoia.xml" don't exist in the driver software in Programs\Euphonics\EuCon\SQRL, copy and paste it here from the folder program folder "/Customize".
3. Go to System Options/Hardware Controller and select "Activate EUCON".
4. Close the dialog. Your controller is now ready to use.

Note

The modes INPUT, DYN, GROUP and MIX are not available in the software.

Controller configuration (Artist control)

Control element	Function	Shift function
SOFT KEY 1	Jog	
SOFT KEY 2	Scrub	
SOFT KEY 3	Loop mode	
SOFT KEY 4	Metronome	
SOFT KEY 5	Sync	
SOFT KEY 6	Position input Apply using ENTER Abort using *	
Numeric keys (0 ~ 9)	Number input	
-		
+		
.	Decimal place	
*	ESC/Cancel	
Enter	Enter	

Control element	Function	Shift function
TRANSPORT KEY 1	Cursor at project start	
TRANSPORT KEY 2	Fast rewind	
TRANSPORT KEY 3	Fast forward	
TRANSPORT KEY 4	Stop	
TRANSPORT KEY 5	Playback	
TRANSPORT KEY 6	Record	
TRANSPORT KEY 7	Loop mode	
WORKSTATION APPLICATION	Switch the current Windows application	Workstation switch
SHIFT	Shift key	

Alesis V49

- First set "V49" as a MIDI input device in Samplitude.
- The Vita instruments supplied react to controllers 21-24 (knobs 1-4).
- Buttons 1+2 (controllers 48+49) can be used for the previous/next preset when the plug-in dialog is open.
- Buttons 3+4 change the keyboard MIDI channel
- Hardware controller setup for simple transport and mixer control (Play/Stop/volume track 1-4)

Note: If you are using V-Editor to configure your keyboard, use "MIDIIN2 (V49)" as a driver.

Frontier AlphaTrack

1. The "AlphaTrack" is supported in Native Mode. For this reason you should set the device to „AlphaTrack Native“ by right-clicking on the "AlphaTrack" button in the taskbar.

2. In Samplitude/Sequoia go to „System Settings/Hardware Controller“, click on the "New" button and select "AlphaTrack" from the list.
3. Now set the right side of the MIDI port to "AlphaTrack".
4. Close the dialog. Your controller is now ready to use.

Controller configuration

Control element	Function	Shift-Label	SHIFT function
Foot switch	Recording		
FADER	Track volume		
ANY SOLO (LED)	Indicates that at least one track in the arrangement is switched to SOLO		
AUTO WRITE (LED)			
AUTO READ (LED)			
REC	Record track		Monitor track
SOLO	Track solo		Global solo
MUTE	Mute track		Global mute
SHIFT	Shift		
PAN	Press first time: Pan mode Second pressing: Active Control Mode (see table 2)		
SEND	AUX mode (see table 2)		
EQ	EQ mode (see table 2)		
Plug-in	Plug-in Mode (see table 2)		
AUTO	Track automation		Copy automation mode to all tracks
F1	Touch automation	F5	Read automation
F2	Latch automation	F6	Automation off
F3	Overwrite automation	F7	Restore
F4	Trim automation	F8	Undo

Control element	Function	Shift-Label	SHIFT function
< TRACK	Previous track	IN	Punch Start Marker
TRACK >	Next track	OUT	Punch End Marker
LOOP	Loop Mode	PUNCH	Punch Marker Mode
FLIP	Mixer	WINDOW	Delete punch marker
<<	Fast rewind	RTZ	Playback marker to project start
>>	Fast forward	END	Playback marker to project end
STOP	Stop	Esc	Cancel
PLAY	Playback	Enter	Enter
RECORD	Recording	MODE	
TOUCHPAD	Positioning		

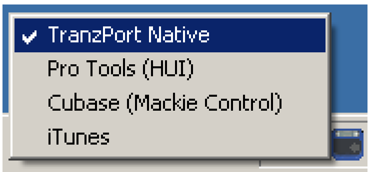
Special modes

Mode	Control element	Turn	Press
Pan mode	KNOB 1	Track selection	
	KNOB 2	To the marker	Set marker
	KNOB 3	Pan	Reset pan
ACTIVE CONTROL MODE	FADER (controls the active element in the mixer, track editor, or plug-in that must be selected with the mouse beforehand)		
	KNOB 1	Pan	
	KNOB 2	Pan	
	KNOB 3	Pan	
AUX mode	KNOB 1	Track selection	
	KNOB 2	AUX level	
	KNOB 3	AUX selection	
EQ mode	KNOB 1	Intensity	

Mode	Control element	Turn	Press
	KNOB 2	Frequency	
	KNOB 3	Quality	
	< TRACK	Previous EQ band	
	TRACK >	Next EQ band	
Plug-in Mode Plug-in slot selection	KNOB 1	Track selection	
	KNOB 2		
	KNOB 3	Slot selection	
	< TRACK	Select plug-in	
	TRACK >	Select plug-in	
Plug-in parameter	KNOB 1	Change parameters	
	KNOB 2	Change parameters	
	KNOB 3	Change parameters	
	< TRACK	Previous parameters	
	TRACK >	Next parameters	

Frontier TranzPort

1. „TranzPort“ is supported in „Native Mode“. For this reason you should set the device to „TranzPort Native“ by right-clicking on the "TranzPort" button in the taskbar.



2. In Samplitude/Sequoia go to "System Settings/Hardware Controller, click the „Add New“ button and select „Frontier_TranzPort (Native)“ from the list.
3. Now,set the right side of the MIDI port to TranzPort.
4. Close the dialog. Your controller is now ready to use.

Controller configuration

Control element	Function	SHIFT function
< TRACK	Previous track	
> TRACK	Next track	
REC	Record track	Monitor track

Control element	Function	SHIFT function
MUTE	Mute track	Global mute
SOLO	Track solo	Global solo
ANY SOLO (LED)	Indicates that at least one track in the arrangement is switched to SOLO	
UNDO	Undo	Restore
IN	Punch Start Marker	
OUT	Punch End Marker	
PUNCH	Punch Marker Mode	Delete punch marker
LOOP	Loop Mode	
SHIFT	Shift	
PREV MARKER	Previous marker	Metronome
ADD MARKER	Insert marker	
NEXT MARKR	Next marker	Scrub mode
<<	Fast rewind	Playback marker to project start
>>	Fast forward	Playback marker to project end
STOP	Stop	Cancel
PLAY	Playback	Enter
RECORD	Recording	
DATA WHEEL	Positioning	Track volume
DATA WHEEL + STOP	Selected range	
Foot switch	Recording	

JLCooper MCS-3800

1. In Samplitude/Sequoia go to "System Settings/Hardware Controller, click the „Add New“ button and select „JLCooper MCS-3800 (Native)“ from the list.
2. Now set up the MIDI ports with the respective inputs and outputs of your MIDI interface on the right side.
3. Close the dialog. Your controller is now ready to use.

Notes

- In Shift Mode the button LEDs are not updated for the status of the commands mapped in "Shift Mode"
- LEDs for Bank3 (time information) and Bank 4 (video window) are inactive.
- Plug-in control is inactive.

Track range

Control element	Function	Modifier	Modifier function
SELECT 1~8	Select track		
AUX 1~8	Record track	SHIFT	Track monitoring 1~8 (ASIO only)
SOLO 1~8	Track solo		
MUTE 1~8	Mute track	SHIFT	Track automation
FADER 1~8 (with fadertouch)	Track volume		

Navigation

Control element	Function
Cursor left	Mixer track 1 to the left
Cursor right	Mixer track 1 to the right
Cursor up	Mixer track 8 to the right
Cursor down	Mixer track 8 to the left

Area to view

Control element	Function
PAGE 1	**Track mode** (see special modes)
PAGE 2	**AUX mode** (see special modes)
PAGE 3	**EQ mode** (see special modes)
PAGE 4	**Active control mode** (see special modes)
PAGE 5	
PAGE 6	
PAGE 7	
PAGE 8	
BANK 1	Mixer
BANK 2	Transport console
BANK 3	Time display
BANK 4	Video window
KNOB 1~5	Change parameters

Functions window

Control element	Function
F1	Punch Marker Mode
F2	Loop
F3	Metronome
F4	Synchronization
F5	
F6	

Control element	Function
F7	
F8	
SHIFT	Shift

Mode range

Control element	Function	Modifier	Modifier function
M1	Read automation	Shift	Automation off
M2	Touch automation		
M3	Latch automation		
M4	Overwrite automation		
M5	Trim automation		

Shuttle range

Control element	Function	Modifier	Modifier function
Shuttle wheel	Positioning	W1	Scrubbing
Rotation ring			
W1	Scrub mode		
W2	Punch Start Marker		
W3	Punch End Marker		
W4	Delete punch marker		
W5			
W6			
W7			

Locator range

Control element	Function
LOCATOR numbers (0 ~ 9)	Entry position: Apply using ENTER or PLAY Abort using STOP
Enter	Apply position
CLEAR / CANCEL	Erase single digits

Control element	Function
+/-	
LAST	
SET LOCATE	
MODE LOCATE	

Transport range

Control element	Function
REWIND	Fast rewind
FAST FORWARD	Fast forward
STOP	Stop
PLAY	Playback
RECORD	Recording

Special modes

Mode	Control element	Function
Track mode	KNOB 1	
	KNOB 2	
	KNOB 3	
	KNOB 4	Track pan
	KNOB 5	To the marker
AUX mode	KNOB 1	AUX level
	KNOB 2	AUX level
	KNOB 3	AUX level
	KNOB 4	AUX level
	KNOB 5	Selection AUX range
EQ mode	KNOB 1	Level EQ band 1
	KNOB 2	Frequency EQ band
	KNOB 3	Quality EQ band
	KNOB 4	

Mode	Control element	Function
	KNOB 5	Band selection
Active control mode	FADER TRACK 1 (controls the active element in the mixer, track editor, or plug-in that must be selected with the mouse beforehand)	

Logic Control

1. In Samplitude go to „System Settings/Hardware Controller“, click on the "Add New" button and select "Logic Control" from the list.
2. Now set up the MIDI ports with the respective inputs and outputs of your MIDI interface.
3. Close the dialog. Your controller is now ready to use.

TRACKS

Control element	Function	Modifier	Modifier function
KNOB 1~8	Pan; Eq; Aux (see **special modes**)		
REC/RDY 1~8	Activate track	SHIFT	Monitor track
SOLO 1~8	Track solo		
MUTE 1~8	Mute track		
SELECT 1~8	Select track		
FADER 1~8 (with fadertouch)	Track volume		
MASTER FADER	Master volume		

ASSIGNMENT

Control element	Function
1	Press first time: Track mode Press second time: Active Control mode (see special modes)

2	Press first time: AUX mode Press second time: AUX mode (phat channel) (see special modes)
PAN/SURROUND	Pan mode (see special modes)
Plug-in	Plug-in Mode (see special modes)
EQ	EQ mode (see special modes)
INSTRUMENT	

FADER BANKS

Control element	Function
< BANK	Mixer track 8 to the left
BANK >	Mixer track 8 to the right
< CHANNEL	Mixer track 1 to the left
CHANNEL >	Mixer track 1 to the right
FLIP	Flip mode: Exchange functions of knobs and faders
GLOBAL VIEW	Mixer

DISPLAY

Control element	Function
NAME/VALUE	
SMTP/BEATS	Timecode switch

FUNCTION BUTTONS

Control element	Function	Modifier	Modifier function
F1-F8	Set / jump to marker 1-8	SHIFT CNTRL	Move markers 1-8 Delete markers 1-8

GLOBAL VIEW

Control element	Function
MIDI TRACKS	

Control element	Function
INPUTS	
AUDIO TRACKS	
AUDIO INSTRUMENT	
AUX	
BUSSES	Select previous object.
OUTPUTS	Select next object
USER	Crossfade Editor

MODIFIERS

Control element	Function
SHIFT	Control
OPTION	Option
CONTROL	
# / ALT	Shift

AUTOMATION

Control element	Function	Modifier	Modifier function
READ/OFF	Read automation	SHIFT	Automation off
WRITE	Overwrite automation		
TRIM	Trim automation		
TOUCH	Touch automation		
LATCH	Latch automation		
GROUP	Copy automation mode to all tracks		

UTILITIES

Control element	Function	Modifier	Modifier function
SAVE	Save		
UNDO	Undo	SHIFT	Repeat
CANCEL	Cancel		
Enter	Enter		

Special modes

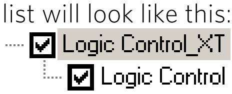
Mode	Control element	Turn	Press
Track mode	KNOB	Pan	Reset
ACTIVE CONTROL MODE	FADER (controls the active element in the mixer, track editor, or plug-in that must be selected with the mouse beforehand)		
Pan mode	KNOB	Pan	Reset
EQ mode	KNOB 1	Level EQ band 1	
	KNOB 2	Level EQ band 2	
	KNOB 3	Level EQ band 3	
	KNOB 4	Level EQ band 4	
	KNOB 5	Frequency EQ band 1	
	KNOB 6	Frequency EQ band 2	
	KNOB 7	Frequency EQ band 3	
	KNOB 8	Frequency EQ band 4	
AUX mode AUX 1~8 selectable using BANK	KNOB 1~8	AUX level	
AUX mode (phat channel) depending on selected track	KNOB 1	AUX 1 level	
	KNOB 2	AUX 2 level	
	KNOB 3	AUX 3 level	
	KNOB 4	AUX 4 level	
	KNOB 5	AUX 5 level	
	KNOB 6	AUX 6 level	
	KNOB 7		
	KNOB 8		
Plug-in Mode	KNOB 1~8		Select plug-in

Mode	Control element	Turn	Press
Plug-in parameter	KNOB 1~8	Change parameters	
	< BANK	Previous parameters	
	BANK >	Next parameters	
Marker mode	REWIND		Previous marker
	FAST FWD		Next marker
Object mode	REWIND		Previous Object
	FAST FWD		Next object

Logic Control XT

1. In Samplitude/Sequoia go to „System Settings/Hardware Controller“, click on the "Add New" button and select "Logic Control XT" from the list.
2. Position Logic Control XT under Logic Control

If you have set up Logic Control XT to the left of Logic Control, the entries in the list will look like this:



3. Now set up the MIDI ports with the respective inputs and outputs of your MIDI interface on the right side.
4. Close the dialog. Your controller is now ready to use.

The functions correspond with the Logic Control.

Note

The number of tracks to be jumped when a bank switch is pressed can be changed in the hardware controller setup options. These settings are made in "Logic Control".

Mackie Control

1. In Samplitude/Sequoia go to „System Settings/Hardware Controller“, click on the "Add New" button and select "Mackie Control" from the list.
2. Now set up the MIDI ports with the respective inputs and outputs of your MIDI interface on the right side.
3. Close the dialog. Your controller is now ready to use.

Note

- All entries are based on MCU Firmware 2.1.2.

TRACKS

Control element	Function	Modifier	Modifier function
KNOB 1~8	Pan; Eq; Aux (see **special modes**)		
REC/RDY 1~8	Activate track	SHIFT	Monitor track
SOLO 1~8	Track solo		
MUTE 1~8	Mute track		
SELECT 1~8	Select track		
FADER 1~8 (with fadertouch)	Track volume		
MASTER FADER	Master volume		

ASSIGNMENT

Control element	Function
1	Press first time: Track mode Press second time: Active Control mode (see special modes)
2	Press first time: AUX mode Press second time: AUX mode (phat channel) (see special modes)
PAN	Pan mode (see special modes)
Plug-ins	Plug-in Mode (see special modes)
EQ	EQ mode (see special modes)
DYN	

FADER BANKS

Control element	Function
< BANK	Mixer track 8 to the left
BANK >	Mixer track 8 to the right
< CHANNEL	Mixer track 1 to the left
CHANNEL >	Mixer track 1 to the right
FLIP	Flip mode: Exchange functions of knobs and faders
EDIT	Mixer

DISPLAY

Control element	Function
NAME/VALUE	
SMTP/BEATS	Timecode switch

FUNCTION BUTTONS

Control element	Function	Modifier	Modifier function
F1-F8	Set / jump to marker 1-8	SHIFT CNTRL	Move markers 1-8 Delete markers 1-8
F9-F13			
F14	Select previous object.		
F15	Select next object		
F16	Crossfade Editor		

MODIFIERS

Control element	Function
CNTRL	Control
OPT	Option
SNAPSHOT	
SHIFT	Shift

AUTOMATION

Control element	Function	Modifier	Modifier function
READ/OFF	Read automation	SHIFT	Automation off
WRITE	Overwrite automation	SHIFT	Copy automation mode to all tracks
UNDO	Undo	SHIFT	Repeat
SAVE	Latch automation		
TOUCH	Touch automation		
REDO	Trim automation		

UTILITIES

Control element	Function
FDR GRP	Cancel
CLR SOLO	Enter
MRKR	
MIXR	Save

TRANSPORT

Control element	Function	Modifier	Modifier function
< FRM	Marker mode (see table 2)		
FRM >	Object mode (see table 2)		
END	Loop mode		
PI	Punch Marker Mode		
PO	Synchronization		
LOOP	Metronome		
HOME	Activate LCD meter by holding "F1", "F2", "F3", or "F4"		
REWIND	Fast rewind	SHIFT	Playback marker to project start

Control element	Function	Modifier	Modifier function
FAST FWD	Fast forward	SHIFT	Playback marker to project end
STOP	Stop		
PLAY	Playback		
RECORD	Recording		
Cursor left	One section left	ZOOM	Horizontal project zoom (-)
Cursor right	One section right	ZOOM	Horizontal project zoom (+)
Cursor up	Next highest track	ZOOM	Vertical project zoom (+)
Cursor down	Next lowest track	ZOOM	Vertical project zoom (-)
ZOOM	Zoom mode		
SCRUB	Scrub mode		
Shuttle wheel	Positioning	SCRUB	Scrubbing

Special modes

Mode	Control element	Turn	Press
Track mode	KNOB	Pan	Reset
ACTIVE CONTROL MODE	FADER (controls the active element in the mixer, track editor, or plug-in that must be selected with the mouse beforehand)		
Pan mode	KNOB	Pan	Reset
EQ mode	KNOB 1	Level EQ band 1	
	KNOB 2	Level EQ band 2	
	KNOB 3	Level EQ band 3	
	KNOB 4	Level EQ band 4	
	KNOB 5	Frequency EQ band 1	

Mode	Control element	Turn	Press
	KNOB 6	Frequency EQ band 2	
	KNOB 7	Frequency EQ band 3	
	KNOB 8	Frequency EQ band 4	
AUX mode AUX 1~8 selectable using BANK	KNOB 1~8	AUX level	
AUX mode (phat channel) depending on selected track	KNOB 1	AUX 1 level	
	KNOB 2	AUX 2 level	
	KNOB 3	AUX 3 level	
	KNOB 4	AUX 4 level	
	KNOB 5	AUX 5 level	
	KNOB 6	AUX 6 level	
	KNOB 7		
	KNOB 8		
Plug-in Mode Plug-in parameter	KNOB 1~8		Select plug-in
	KNOB 1~8	Change parameters	
	< BANK	Previous parameters	
	BANK >	Next parameters	
Marker mode	REWIND		Previous marker
	FAST FWD		Next marker
Object mode	REWIND		Previous Object
	FAST FWD		Next object

Mackie Control XT

1. In Samplitude/Sequoia go to „System Settings/Hardware Controller“, click on the "Add New" button and select "Mackie Control XT" from the list.
2. Position Mackie Control XT under Mackie Control.

If you have set up Mackie Control XT to the left of Mackie Control, the entries in the list will look like this:

☑ Mackie Control_XT
  ☑ Mackie Control

3. Now set up the MIDI ports with the respective inputs and outputs of your MIDI interface on the right side.
4. Close the dialog. Your controller is now ready to use.

The functions correspond with the Mackie Control.

Note

> The number of tracks to be jumped when a bank switch is pressed can be changed in the hardware controller setup options. These settings are available in Mackie Control.

PreSonus FaderPort

1. In Samplitude/Sequoia go to "System Settings/Hardware Controller, click the „Add New“ button and select „PreSonus FaderPort (Native)“ from the list.
2. Now set the MIDI ports on the right side to USB audio device.
3. Close the dialog. Your controller is now ready to use.

Notes

> - The foot switch is functional starting with the hardware controller firmware 1.2.
> - To map control elements for the "FaderPort", enter the response values to the controller for learning so that the button LEDs also function. To do this

copy and paste the value from the last column of the assignment table (view page 522) of the corresponding control element into the field „Send MIDI Data“.

Controller configuration

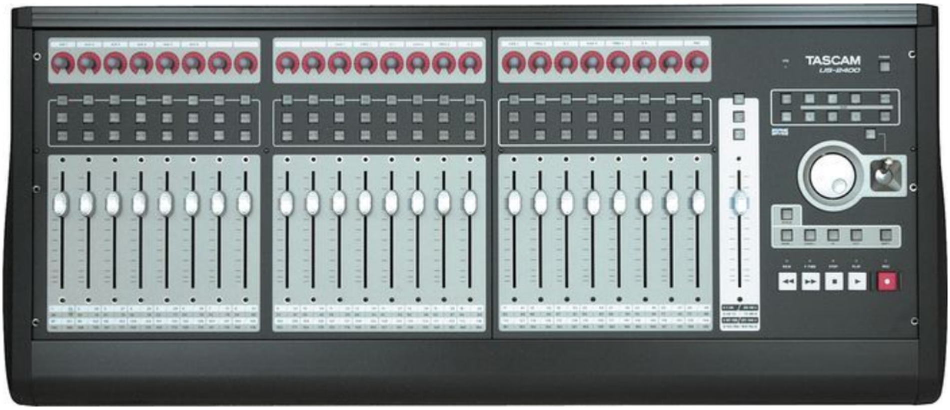
Control element (SHIFT)	Function	SHIFT function	MIDI Command	Return value for LED
FADER	Track volume			
PAN	Track pan			
MUTE	Mute track	Global mute	A01200	A01501
SOLO	Track solo	Global solo	A01100	A01601
REC	Record track	Monitor track (ASIO only)	A01000	A01701
<	Previous Track/Bank		A01300	A01401
BANK	"<" and ">" switch bank		A01400	A01301
>	Next Track/Bank		A01500	A01201
OUTPUT	ACTIVE CONTROL MODE (see special modes (view page 523))		A01600	A01101
READ	Read automation		A00A00	A00D01
WRITE	Overwrite automation	Latch automation	A00900	A00E01
TOUCH	Touch automation	Trim automation	A00800	A00F01
OFF	Automation off	Copy automation mode to all tracks	A01700	A01001
MIX	Mixer		A00B00	A00C01
PROJ	Focus on arranger		A00C00	A00B01

Control element (SHIFT)	Function	SHIFT function	MIDI Command	Return value for LED
TRNS	Transport console		A00D00	A00A01
UNDO (REDO)	Undo	Restore	A00E00	A00901
SHIFT	Shift		A00200	A00501
PUNCH (PREV)	Punch Marker Mode	Previous marker	A00100	A00601
USER (NEXT)	Metronome	Next marker	A00000	A00701
LOOP (MARK)	Loop Mode	Insert marker	A00f00	A00801
REWIND (START)	Fast rewind	Playback marker to project start	A00300	A00401
FAST FORWARD (END)	Fast forward	Playback marker to project end	A00400	A00301
STOP	Stop		A00500	A00201
PLAY	Playback		A00600	A00101
RECORD	Recording		A00700	A00001
FOOT SWITCH	Recording		A07e00	

Special modes

Mode	Control element
ACTIVE CONTROL MODE	FADER (controls the active element in the mixer, track editor, or plug-in that needs to be selected with the mouse beforehand)

Tascam US-2400



1. The "US-2400" is supported in "Native Mode". Use the most current firmware. Start the device with the "Power" button while holding down the "SEL" (above the master fader) and "CHAN" buttons. This puts the device in "Native Mode".
2. In Samplitude/Sequoia go to "System Settings/Hardware Controller, click the „Add New“ button and select „Tascam US-2400 (Native)“ from the list.
3. Now set the MIDI ports on the right side to USB audio device.
4. Close the dialog. Your controller is now ready to use.

Notes

- The number of tracks to be jumped when a bank switch is pressed can be changed in the hardware controller setup options in the program.
- When using the TASCAM US-2400 as an exclusive recording panel, it's recommended that you relearn the SEL buttons as RecordReady.

Track range

Control element	Function	Shift function
KNOB 1~24	Pan, EQ, AUX (see table 2)	
SEL 1~24	Select track	Track automation
SOLO 1~24	Track solo	Record track
MUTE 1~24	Mute track	Monitor track (ASIO only)
FADER 1~24 (with fadertouch)	Track volume	

Control element	Function	Shift function
SEL (master)		Copy automation mode to all tracks
CLR SOLO	Global solo	Global record
FLIP	Global mute	Flip mode: Exchange functions of knobs and faders
MASTER FADER	Master volume	

Navigation

Control element	Function	Shift function
CHAN	**EQ mode** (see special modes)	
PAN	**Pan mode** (see special modes)	
AUX 1	KNOB 1~24 to AUX 1	Read automation
AUX 2	KNOB 1~24 to AUX 2	Touch automation
AUX 3	KNOB 1~24 to AUX 3	Latch automation
AUX 4	KNOB 1~24 to AUX 4	Overwrite automation
AUX 5	KNOB 1~24 to AUX 5	Trim automation
AUX 6	KNOB 1~24 to AUX 6	Automation off
METER	Activate level meter in knob display	
F-KEY	Active control mode (see special modes)	
NULL		
Shuttle wheel	Positioning	
Shuttle wheel + STOP	Select range	
SCRUB	Scrub mode	

Control element	Function	Shift function
BANK	Mixer track 24 to the left	Mixer track 1 to the left
BANK+	Mixer track 24 to the right	Mixer track 1 to the right
IN	Loop Mode	
OUT	Metronome	
SHIFT	Shift	
REW	Fast rewind	Playback marker to project start
FFW	Fast forward	Playback marker to project end
STOP	Stop	Cancel
PLAY	Playback	Enter
REC	Recording	
Foot switch	Recording	

Special modes

Mode	Control element	Function
Pan mode	KNOB 1 ~ 24	Panorama
EQ mode (1 press)	KNOB 1	Level EQ band 1
	KNOB 2	Frequency EQ band 1
	KNOB 3	Quality EQ band 1
	KNOB 4	
	KNOB 5	Level EQ band 2
	KNOB 6	Frequency EQ band 2
	KNOB 7	Quality EQ band 2
	KNOB 8	
	KNOB 9	Level EQ band 3
	KNOB 10	Frequency EQ band 3
	KNOB 11	Quality EQ band 3
	KNOB 12	

Mode	Control element	Function
	KNOB 13	Level EQ band 4
	KNOB 14	Frequency EQ band 4
	KNOB 15	Quality EQ band 4
	KNOB 16	
EQ mode (2 press)	KNOB 1	Level EQ band 1
	KNOB 2	Level EQ band 2
	KNOB 3	Level EQ band 3
	KNOB 4	Level EQ band 4
	KNOB 5	Frequency EQ band 1
	KNOB 6	Frequency EQ band 2
	KNOB 7	Frequency EQ band 3
	KNOB 8	Frequency EQ band 4
	KNOB 9	Quality EQ band 1
	KNOB 10	Quality EQ band 2
	KNOB 11	Quality EQ band 3
	KNOB 12	Quality EQ band 4
ACTIVE CONTROL mode	FADER TRACK 1 (controls the active element in the mixer, track editor, or plug-in that must be selected with the mouse beforehand)	

Tascam FW-1884

1. Open the FW-1884 Control Panel (see TASCAM documentation) and set the "Control Protocol" to "Mackie Control Emulation".
2. In Samplitude/Sequoia go to "System Settings/Hardware Controller, click the „Add New“ button and select „Tascam FW-1884 (Mackie)“ from the list.
3. Now set the MIDI ports on the right side to "FW 1884 Control".
4. Close the dialog. Your controller is now ready to use.

Notes

The following elements do not send MIDI values and cannot be learned:

- PFL, HIGH, HI-MID, LOW-MID, LOW, GAIN, FREQ, Q, COMPUTER, MIDI CTRL, MON, CLOCK, ROUTE
- All elements with CTRL configuration can be mapped in the control panel without providing a modifier because the CTRL configuration already takes place in the controller and sends separate IDs.
- All elements with a CTRL configuration can also be set up with SHIFT-CTRL. In the control panel, set the modifier to "Shift" (since the CTRL configuration takes place in the controller already), next press CTRL + the element and learn.
- LED for AUX-elements is not functioning properly.
- FLIP LED is not functioning.
- The SoftLCD application that was installed with the FW-1884 drivers can be used to display track names and other typical Mackie Control information

Encoder

Control element	Function	Shift function
FLIP	Flip mode: KNOBS and FADERS swap functions	
PAN	Pan mode	
AUX 1~6	Aux select	
AUX 7, 8		

Shortcuts, control panel

Control element	Function	Modifier	Modifier function
SAVE/F1	Lock project		
REVERT/F2	Global mute	SHIFT	Global recording
ALLSAFE/F3			
CLR SOLO/F4	Global solo		
MARKER/F5	Metronome		
LOOP/F6	Loop Mode		
CUT	Split object	SHIFT	Split Objects with Alternate (Linear) Fade
DEL	Delete	SHIFT	Ripple delete
COPY	Copy	CTRL	Duplicating and moving
PASTE	Insert		
ALT/CMD			
UNDO	Undo	SHIFT	Repeat
SHIFT	Shift		
CTRL	Ctrl		

Track range

Control element	Function	Modifier	Modifier function
ENCODERS 1-8	Send pan/aux		
SEL 1-8	Select track	CTRL	Monitor track
SOLO 1-8	Track solo		
MUTE 1-8	Mute track		

Control element	Function	Modifier	Modifier function
FADER 1-8	Track volume		
FADER MASTER	Master volume		
REC	Arm track with SEL 1-8		

Navigation

Control element	Function	Modifier	Modifier function
F7	Automation OFF		
F8	Transfer automation mode to all tracks		
F9	Trim automation		
F10	Mixer		
READ	Read automation		
WRIT	Overwrite automation		
TCH	Touch automation		
LATCH	Latch automation		
SIITL	Scrub mode	SHIFT	Zoom mode
WHEEL	Jog/Scrub		
CURSOR UP	Previous track	SHIFT	Zoom project vert.(+)
CURSOR DOWN	Next track	SHIFT	Zoom project vert.(-)
CURSOR LEFT	Timeline positioning	SHIFT	Zoom project horiz.(-)
CURSOR RIGHT	Timeline positioning	SHIFT	Zoom project horiz.(+)
BANK LEFT	Previous bank	SHIFT	Mixer channel 1 left
BANK RIGHT	Next bank	SHIFT	Mixer channel 1 right
NUDGE LEFT	Move object left		
NUDGE RIGHT	Move object right		
LOCATE LEFT	Previous marker	SHIFT	Previous Object
LOCATE RIGHT	Next marker	SHIFT	Next object
SET	Create marker (auto #)	SHIFT	Delete marker
IN	Set punch in marker	SHIFT	Delete punch marker
OUT	Set punch-out marker		

Control element	Function	Modifier	Modifier function
REW	Fast rewind	SHIFT	Playback marker to project start (Return to Zero)
FFW	Fast forward	SHIFT	Playback marker to project end (Go to End)
STOP	Stop	SHIFT	Cancel
PLAY	Playback (second press: stop at position)	SHIFT	Enter
REC	Recording	CTRL	Punch mode on/off

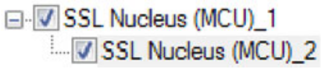
SSL Nucleus

Set up SSL Nucleus

1. In the "Nucleus Remote" configuration tool of the SSL Nucleus set the "Samplitude Default" profile.
   - If this profile is not displayed, you can add it through "Edit Profiles" > "Restore". The file can be found in the Samplitude/Sequoia program folder under "Customize\SSL Nucleus\".
2. The "SSL Nucleus" is connected with the computer through ethernet:
   - SSL provides a help program for this („ipMIDI Ethernet Midi Ports“) that supplies the operating system with virtual MIDI ports.
   - Configure this program so that there are at least 2 ports available.
   - You should also deactivate the "Loop Back" option.
   - Up to three DAWs can be controlled with the Nucleus. For this the virtual MIDI ports are assigned as follows:
     DAW 1: ipMIDI ports 1 and 2

DAW 2: ipMIDI ports 3 and 4
DAW 3: ipMIDI ports 5 and 6

3. Go to „System Settings/Hardware Controller“ in Samplitude and click on „Add New“ to select „SSL Nucleus (MCU)_1“ from the list.
4. You can also add „SSL Nucleus (MCU)_2“ in exactly the same way.
5. Position „SSL Nucleus (MCU)_2“ indented under „SSL Nucleus (MCU)_1“.



6. Now set MIDI ports for both entries:
   - „SSL Nucleus (MCU)_1": „1. Ethernet MIDI"
   - „SSL Nucleus (MCU)_2": „2. Ethernet MIDI" (if Samplitude/Sequoia is DAW 1).
7. Close the dialog. Your controller is now ready to use.

Various Modes

Control element	Function
INSTRUMENT	
Plug-in	Plug-in Mode (see special modes (view page 536))
EQ	1. Press: EQ Mode 1 (Band) 2. Press: EQ Mode 2 (Type) (see special modes (view page 536))
PAN	
SEND	1. Press: AUX mode (phat channel) 2. Press: AUX Mode (Track Mode) (see special modes (view page 536))
TRACK	1. Press: Track mode 2. Press: Active control mode (see special modes (view page 536))

Control element	Function
USER1	User1 Mode (view page 535)
USER2	
PARAM	Mixer Window
EQ	
REC	Record Mode (view page 534)
FLIP	Flip Mode: Knobs and faders exchange their functions.
DYN	
AUTO	Automation mode (view page 535)

General Control Elements

Control element	Function	Modifier	Modifier function
PARAM	Mixer Window		
< BANK	Bank to the left		
BANK >	Bank to the right		
< CHANNEL	One channel to the left		
CHANNEL >	One channel to the right		
FLIP	Flip mode: Exchange functions of knobs and faders		
SHIFT	Modifier Shift		
OPT	Modifier Option		
CNTL	Modifier Control		
ALT			
DAW 2	Control of second DAW (Assignment in Nucleus Remote)		
Enter	Enter		
SAVE	Save		
F1			
DAW 3	Control of third DAW (Assignment in Nucleus Remote)		

Control element	Function	Modifier	Modifier function
Esc	Cancel		
UNDO	Undo	SHIFT	Redo
F2			

Record Mode

Control element	Function	Modifier	Modifier function
SEL 1 ~ 16	Activate track	SHIFT	Track monitoring

Tracks

Control element	Function	Modifier	Modifier function
KNOB 1~16	Pan; Eq; Aux; Plug-in parameters (see Special Modes (view page 536))		
CUT 1~16	Mute track		
SOLO 1~16	Track solo		
SEL 1~16	Select track		
FADER 1~16 (with fader touch)	Track volume	SHIFT	Touch fader: Reset to 0 dB

Navigation

Control element	Function	Function in Zoom-Mode
MODE	Zoom mode	
Arrow up	Select previous track	Zoom in arranger view vertically
Arrow down	Select previous track	Zoom out arranger view vertically
Arrow left	Move arranger view forward on the timeline	Zoom in arranger view horizontally (along the timeline)
Arrow right	Move arranger view backward on the timeline	Zoom out arranger view horizontally (along the timeline)

Transport

Control element	Function	Modifier	Modifier function
PREVIOUS	Playback marker to previous object	SHIFT	To project start
NEXT	Playback marker to next object	SHIFT	To project end
CYCLE	Loop Mode		
REWIND <<	Fast rewind		
FFWD >>	Fast forward		
STOP	Stop		
PLAY	Playback		
RECORD	Recording		
Shuttle wheel	Positioning	SCRUB	Scrubbing
Wheel Button	Scrub mode		

Automation

Control element	Function	Modifier	Modifier function
Rd/Off	Read automation	SHIFT	Automation off
Write	Overwrite automation	SHIFT	Copies automation mode to all tracks
Touch	Touch automation		
Latch	Latch automation		

User1 Mode

Control element	Function	Modifier	Modifier function
Mixer	Mixer window		
Transp	Transport window		
Markers	Previous/Next marker with FFWD/REWIND		
Nudge	Previous/Next object with FFWD/REWIND		
SMPTE	Timecode switch		
Click	Metronome on/off		

Control element	Function	Modifier	Modifier function
1	Set/Jump to Marker 1	SHIFT CNTL	Move Marker 1 Delete Marker 1
2	Set/Jump to Marker 2	SHIFT CNTL	Move Marker 2 Delete Marker 2
3	Set/Jump to Marker 3	SHIFT CNTL	Move Marker 3 Delete Marker 3
4	Set/Jump to Marker 4	SHIFT CNTL	Move Marker 4 Delete Marker 4
5	Set/Jump to Marker 5	SHIFT CNTL	Move Marker 5 Delete Marker 5
6	Set/Jump to Marker 6	SHIFT CNTL	Move Marker 6 Delete Marker 6

Special modes

Mode	Control element	Turn	Press
Plug-in Mode Plug-in Parameter	KNOB 1~6	–	Select plug-in
	KNOB 1~16	Change parameters	–
	< BANK	Previous parameters	–
	BANK >	Next parameters	–
EQ Mode 1 Banks <,> are not available in this mode	KNOB 1	Level EQ band 1	–
	KNOB 2	Frequency EQ band 1	–
	KNOB 3	Quality EQ band 1	–
	KNOB 4	Level EQ band 2	–
	KNOB 5	Frequency EQ band 2	–
	KNOB 6	Quality EQ band 2	–
	KNOB 7	Level EQ band 3	–
	KNOB 8	Frequency EQ band 3	–
	KNOB 9	Quality EQ band 3	–
	KNOB 10	Level EQ band 4	–

Mode	Control element	Turn	Press
	KNOB 11	Frequency EQ band 4	–
	KNOB 12	Quality EQ band 4	–
EQ Mode 2 Banks <,> are not available in this mode	KNOB 1	Level EQ band 1	–
	KNOB 2	Level EQ band 2	–
	KNOB 3	Level EQ band 3	–
	KNOB 4	Level EQ band 4	–
	KNOB 5	Frequency EQ band 1	–
	KNOB 6	Frequency EQ band 2	–
	KNOB 7	Frequency EQ band 3	–
	KNOB 8	Frequency EQ band 4	–
	KNOB 9	Quality EQ band 1	–
	KNOB 10	Quality EQ band 2	–
	KNOB 11	Quality EQ band 3	–
	KNOB 12	Quality EQ band 4	–
AUX Mode (Phat Channel) depending on active track	KNOB 1	AUX 1 Level	–
	KNOB 2	AUX 2 Level	–
	KNOB 3	AUX 3 Level	–
	KNOB 4	AUX 4 Level	–
	KNOB 5	AUX 5 Level	–
	KNOB 6	AUX 6 Level	–
	KNOB 7	–	–
	KNOB 8	–	–

Mode	Control element	Turn	Press
AUX Mode (Track Mode) For example, on all knobs there is AUX 1, selectable through Ban k< and/or Bank >	KNOB 1~16	AUX level	–
Track mode	KNOB	Pan	Reset
ACTIVE CONTROL MODE	FADER 1 (controls the active element in the mixer, track editor, or plug-in that was selected with the mouse beforehand)	–	–
Marker mode	REWIND <<	–	Previous marker
	FFWD >>	–	Next marker
Object mode	REWIND <<	–	Previous Object
	FFWD >>	–	Next object

Preset keyboard shortcuts

On the other hand, you may redefine this keyboard shortcut whenever you like via "Menu File > Program preferences > Edit Keyboard Shortcuts and Menu (view page 626)". Change, remove, create keyboard shortcuts. Clicking the "All" dialog opens an overview of all of the shortcuts defined there. Shortcuts in brackets (<>) cannot be changed.

Note: Pressing the key combination "Alt + Shift" may cause Windows to switch keyboard languages. In order to avoid this, the keyboard shortcut for switching the input scheme can be deactivated. In the Windows control panel, open the "Country and language" options. Navigate to the "Languages" tab and then to the "Details" button to access the settings for the standard input scheme. Press "Keyboard" to access the advanced keyboard settings. Open the dialog "Change keyboard combination" and place a check mark next to "Change input scheme" and "Remove keyboard layout" to hinder the input scheme from changing in the future.

File menu

New Virtual Project (VIP)...	E

Open/Import/Save

Virtual project (*.vip)...	O
Load MIDI (*.MID) file	Shift + M
Open HD Wave Project (*.hdp)	Shift + L
Load audio file	W
Save project	Ctrl + S
Save project as...	Shift + S
Save copy as...	Ctrl + Alt + S
Import audio...	Ctrl + I
Export WAV	Ctrl + E

Project properties/Program settings

Snap and grid setup...	Ctrl + Shift + ´, I
Mixer settings...	Ctrl + Shift + M
Project status...	Shift + I

System options...	Y
Playback parameters...	P
Synchronization active	G
Synchronization settings...	Shift + G
Project display	Shift + Tab

Edit menu

Undo	Ctrl + Z
Restore	Ctrl + Y
Copy	C / Ctrl+C / Ctrl+Insert
Copy to...	Shift + C
Copy + clear	Ctrl + Alt + C
Cut into clip	X / Ctrl + X / Shift + Del
Cut with time/ripple	Ctrl + Alt + X
Split objects	T
Undo object split	Ctrl + Alt + T
Paste / Insert clip	V / Ctrl + V / Shift + Insert
Insert from clips with time/ripple	Ctrl + Alt + V
Overwrite with clip	Alt + V / Insert
Delete	Del
Delete with time/ripple	Ctrl + Del
Replace with silence	Alt + Del
Metronome active	Ctrl + #
Crossfade-Editor	Strg + F
Time Display Field 1...5	Alt + Numpad 1 ...5

Range

Range all	A

Manipulate range

Move range start to left	Alt + Numeric Pad '/'
Move range start to right	Alt + Numeric Pad '*'
Move range end to left	Shift + Left arrow / Alt + Numeric Pad -
Move range end to right	Shift + Right arrow / Alt + Numeric Pad '+'
Fold range to left	Ctrl + Shift + Left arrow
Flip Range right	Ctrl + Shift + Right arrow
Beginning of Range -> 0	Ctrl + Page up
Beginning of Range <- 0	Shift + Page up
End of Range -> 0	Ctrl + Page down
End of Range <- 0	Shift + Page down
Range start to left marker	Shift + F2
Range end to right marker	Shift + F3
Split range	B
Store range	Alt + F2 - F10 (except Alt + F4)
Other Range...	Alt + F11
Get range	Ctrl + F2 - F10
Get range length	Ctrl + Shift + F2 - F10
Saver marker	Shift + 0-9
Marker with name...	?
Markers with auto number	Shift+#
Place marker at recording position	Alt + ?
Get markers	1 - 9, 0
Recall last range	Shift + Back

Track menu

Track input mono	Alt + N
Maximize track	Alt + Enter
Freeze track	Alt + Shift + F

Unfreeze track	Alt + Shift + U
Track Options...	Alt + I
Next revolver track	Alt+Pg Dn
Previous revolver track	Alt+Pg Up
Edit Volume	Ctrl + Shift + K
Edit Pan	Ctrl + Shift + P
Track EQ	Ctrl + Shift + F

Track properties

Mute	Alt + M
Mute / Inactive	Ctrl + Alt + M
Solo	Alt + S
Solo exclusive	Shift + Alt + S
Record ready	Alt + R
Monitoring	Ctrl + Alt + Shift + F
Lock	Alt + L
Volume curve active	Alt + K
Pan curve active	Alt + P
Track phase invert	H
Activate next track	Alt + Arrow down
Activate previous track	Alt + Arrow up

Object menu

Object editor...	Ctrl + O
Set hotspot	Ctrl + H
Take manager	Ctrl + Alt + Shift + T
Object name	Ctrl + N
Freeze object	Ctrl + Alt + F
Unfreeze object	Ctrl + Alt + U
Time stretch / Pitch shift patcher	J

Edit objects

Duplicate and move	Ctrl + D
Build loop object	Ctrl + L
Split objects	T
Undo object split	Ctrl + Alt + T
Trim objects	Ctrl + T
Glue objects	Ctrl + Alt + G
Mute objects	Ctrl + M

Select objects / Groups

Select all objects	Ctrl + A
Object lasso	Ctrl + Alt + L
Select previous object	< / Ctrl + Alt + Q
Select next object	> / Ctrl + Alt +W
Select to previous object	Ctrl + Alt + Shift + Q
Select to next object	Ctrl + Alt + Shift + W
Deselecting objects	Ctrl + Shift + A
Group objects	Ctrl + G
Ungroup objects	Ctrl + U
Temporarily exclude object from group	Ctrl + Shift + U

Move or edit object/crossfade

Arrange objects	Ctrl + Alt + Shift + A
Object hotspot to play cursor position	Ctrl + Alt + P
Object to original position	Ctrl + Alt + O

Object step size 1

Move (left) Object Left	Ctrl + 1
Move right Object Left	Alt + 1
Move (left) Object Right	Ctrl + 2
Move right Object Right	Alt + 2
Move Object(s) Left	Ctrl + Alt + 1

Move Object(s) Right	Ctrl + Alt + 2
Move Object Start Left	Ctrl + 3
Move Object Start Right	Ctrl + 4
Move Object End Left	Alt + 3
Move Object End Right	Alt + 4
Move Crossfade Left	Ctrl + Alt + 3
Move Crossfade Right	Ctrl + Alt + 4
Move Object Start Offset Left	Ctrl + 5
Move Object Start Offset Right	Ctrl + 6
Move Object End Offset Left	Alt + 5
Move Object End Offset 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
Move (left) Object Content Left	Ctrl + 9
Move (left) Object Content Right	Ctrl + 0
Move right Object Content Left	Alt + 9
Move right Object Content Right	Alt + 0
Move Object(s) Content Left	Ctrl + Alt + 9
Move Object(s) Content Right	Ctrl + Alt + 0

Object step size 2

Move (left) Object Left	Ctrl + Shift + 1
Move right Object Left	Alt + Shift + 1
Move (left) Object Right	Ctrl + Shift + 2
Move right Object Right	Alt + Shift + 2
Move Object(s) Left	Ctrl + Alt + Shift + 1
Move Object(s) Right	Ctrl + Alt + Shift + 2

Move Object Start Left	Ctrl + Shift + 3
Move Object Start Right	Ctrl + Shift + 4
Move Object End Left	Alt + Shift + 3
Move Object End Right	Alt + Shift + 4
Move Crossfade Left	Ctrl + Alt + Shift + 3
Move Crossfade Right	Ctrl + Alt + Shift + 4
Move Object Start Offset Left	Ctrl + Shift + 5
Move Object Start Offset Right	Ctrl + Shift + 6
Move Object End Offset Left	Alt + Shift + 5
Move Object End Offset Right	Alt + Shift + 6
Increase left volume	Ctrl + Shift + 8
Decrease left volume	Ctrl + Shift + 7
Increase right volume	Alt + Shift + 8
Decrease right volume	Alt + Shift + 7
Increase volume	Ctrl + Alt + Shift + 8
Decrease volume	Ctrl + Alt + Shift + 7
Move (left) Object Content Left	Ctrl + Shift + 9
Move (left) Object Content Right	Ctrl + Shift + 0
Move right Object Content Left	Alt + Shift + 9
Move right Object Content Right	Alt + Shift + 0
Move Object(s) Content Left	Ctrl + Alt + Shift + 9
Move Object(s) Content Right	Ctrl + Alt + Shift + 0

Play

Play / Play loop	Space bar
Play with preload	Shift + Spacebar
Play into Loop	Shift + P
Play only selected objects	Ctrl + space bar
Stop and go to current position	Numeric keypad , / Pause
Playback parameters	P

Record	R
Recording Options	Shift + R
Input Monitoring	Alt + Shift + M

Play cut

Play to cut start	F5
Play from cut start	F6
Play to cut end	F7
Play from cut end	F8
Play over cut	F4

Playback mode

Scrubbing active	Alt + Shift + Arrow down
Scrubbing left	Alt + Shift + Arrow left
Scrubbing right	Alt + Shift + Arrow right

Move play cursor

Play cursor to start	Home
Play cursor to end	End
Left move in page mode	Arrow left
Left move in scroll mode	Alt + Arrow left
Right move in page mode	Arrow right
Right move in scroll mode	Alt + Arrow right
Object border left	Ctrl + Q
Object border right	Ctrl + W
Marker left	F2 / Alt + Q
Marker right	F3 / Alt + W
Section to play cursor/last stop position	Ctrl + Alt + ,
Repeat last position(s)	Backspace

Markers

Marker with name...	?
Markers with auto numeration	Shift + `
Place marker at recording position	Alt + ?
Set markers 1-10	Shift + 1...0
Jump to markers 1-10	1..0

Effects menu

Normalize...	Shift + N
Normalize (quick access)	N
Elastic Audio	Ctrl + Shift + E

CD/DVD menu

Set track	Ctrl + Alt + I
Remove all indices	Ctrl + Alt + Shift + I
CD track options	Ctrl + Alt + Shift + D

View menu

Mixer	M
Transport console	Ctrl + Shift + T
Time display	Ctrl + Shift + Z
Visualization	Ctrl + Alt + Shift + V
Track editor	Ctrl + Alt + Shift + E
Keyboard	Ctrl + Alt + Shift + K
Autoscroll	Scroll
Soft Autoscroll	Shift + Scroll
Tile	Enter
Restore	Shift + Enter

Manager

File manager	Ctrl + Shift + B
Object manager	Ctrl + Shift + O
Track manager	Ctrl + Shift + S
Marker manager	Ctrl + Alt + Shift + M
Range manager	Ctrl + Alt + Shift + B
Take manager	Ctrl + Alt + Shift + T
VST instruments manager	Ctrl + Shift + I
Routing manager	Ctrl + Alt + Shift + R

Sections/Grid view/VIP display

Activate next section	Page down
Activate previous section	Page up
Grid	´
Snap active	Ctrl + ´
Snap / Grid Setup...	Ctrl + Shift + ´, I
Define VIP display...	Shift + Tab
Toggle between mode 1 / 2	Tab

Save position and zoom level

1	Ctrl + NumPad 1
2	Ctrl + NumPad 2
3	Ctrl + NumPad 3

Save zoom level

1	Ctrl + NumPad 4
2	Ctrl + NumPad 5
3	Ctrl + NumPad 6

Get position and zoom level

1	Number block "1"
2	Number block "2"
3	Number block "3"

Get zoom level

1	Number block "4"
2	Number block "5"
3	Number block "6"

Horizontal

Half section left	Ctrl + Alt + Arrow left
Half section right	Ctrl + Alt + Arrow right
Section to play cursor/last stop position	Ctrl + Alt + ,
Section to range start	Ctrl + Alt + B / Ctrl + Shift + Page up
Section to range end	Ctrl + Alt + N / Ctrl + Shift + Page down
Zoom In	Arrow up / Ctrl + Arrow right
Zoom out	Arrow down / Ctrl + Arrow right
Show all	Ctrl + Alt + Arrow up
Zoom to range	Ctrl + Alt + Arrow down

Vertical

Half section up	Shift + Arrow up
Half section down	Shift + Arrow down
Zoom into waveform	Ctrl + Arrow up
Zoom out of waveform	Ctrl + Arrow down
Zoom-in volume automation	Ctrl + Shift + Arrow up
Zoom-out volume automation	Ctrl + Shift + Arrow down

Window

Close all windows	Ctrl + H
Activate project window	Ctrl + P
Activate docking window	Ctrl + B
Close current docker tab	Ctrl + F4
Next docker tab	Ctrl + Tab

Help menu

Help	F1
Context help	Shift + F1

Mouse

Middle-click	Play Start/Stop
Mouse wheel	Vertical scrolling in VIP (track)
Shift + Mouse wheel	Horizontal scrolling
Ctrl + Mouse wheel	Vertical zooming
Alt + Mouse wheel	Wave Zooming
Ctrl + Alt + Mouse wheel	Horizontal and vertical zooming
Ctrl + Shift + Mouse wheel	Horizontal zooming

Glossary

A

Active section

If you wish to use zoom commands with the position buttons or the menu commands (position bar) on a section, it's necessary to specify the section beforehand for which the corresponding positioning is required. This happens by clicking on the right or lower scroll bars of the section.

Detailed information about this is available under:

"View -> Sections (view page 1032)"

Edit menu "Range" > Split Range (view page 659)"

Audio markers

Markers in wave projects are saved in the audio file (*.wav) as audio markers and are available in this form in other applications as well. Audio markers are coupled directly with the audio material and visible along the upper edge of an audio object. The purpose of audio markers is to mark positions within the audio material so that the selections remain independent of their placement within the virtual project.

Audio markers may also be made visible in the display options ("Shift + Tab") in the "Objects" area by marking "Audio markers" with a check.

The markers displayed in a virtual project's object are identical to the markers in the associated wave project. If you set new project markers in a wave project (this happens automatically while recording a take), then all audio markers in all associated objects of the virtual project will be visible at the same position in the audio material.

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

Automation

Automation can be used to change the envelope of certain parameters in Samplitude such as panorama, volume, surround and AUX parameters, MIDI controllers, and VST parameters. Changes can be displayed with the help of an automation curve.

There is a separate mouse mode available for editing automation curves.

AUX bus

An AUX bus is a track featuring all the abilities of a normal track, i.e. it can include track effects, and panorama and level can be automated. It has one input and one output. The most important difference is that the audio signal is diverted to the bus of any track with a lower track number than the AUX bus. This is done via the AUX send faders in the mixer.

AUX buses are typically used for effects like reverb or echo, which may be equally required for various tracks, but in different amounts. Objects may also be routed to AUX buses.

C

Clip for audio files

The clip is a buffer or a clipboard where samples from wave projects can be copied into or from which you can paste samples into wave projects. The content of the clip can also be mixed with the data of an audio file. The clip always assumes the properties of the project where the data originated (bitmap resolution, sample rate, and mono, LR or stereo mode). The clip is presented on the screen by a window with Clip in the title.

The clip window is normally hidden; however, you can make it visible with the command "View > Window> Audio file window > Iconize all Wave projects". Otherwise, the clip is a project like any other, i. e. you can play it, edit it, and save it with another name.

Context menu

Nearly all parts of a VIP and the mixer include a context menu that is opened via the right mouse button. It is called a context menu because different menus appear that are always adapted to the current context, i.e. depending on the VIP or mixer area that is clicked.

Crossfade

Crossfading two objects on one track within the Virtual Project (VIP). A standard crossfade can be created automatically when cutting to avoid clicking ("Auto crossfade" mode). Crossfades can also be edited in Samplitude in detail with the crossfade editor.

D

Destructive editing

In this mode, wave projects are edited directly on the hard drive. This means that changes to the audio material which occur acoustically and visually in the project window are also included in the audio file. The file does not have to be saved after editing, since editing of the audio data can be performed directly on the hard drive. Of course, undo steps are also available, but these are only available for as long as the file is open. After that the changes are permanent!

Note: Destructive editing can be used in virtual projects if offline effects are applied to selected objects or when working with wave editing in "Destructive" mode.

F

Fade

An audio transition. These may be fade-ins (volume increase) or fade-outs (volume decrease). You can also fade between objects in the VIP (crossfade).

H

Handles

Handles are the five little rectangles that appear on the outline of a selected object.

The **length handles** are on the bottom left and right corners of the object. These may be used to adjust the length of the object.

Fade handles: Located on the top right and top left. These can be used to softly fade an object in or out.

Volume handles: These are located in the center. Use these handles to adjust the track volume.

Hybrid Engine and economy tracks

Very small audio buffers ("Low latency" mode) increase the CPU strain of the playback engine. For optimal use of the PC resources, we recommend using the mixer tracks in "Low latency" mode only. This includes:

- Monitor tracks
- Tracks with VSTis

Tracks that play hard disk content (MIDI or audio files) may be taken out of "Low latency" mode in the Hybrid Engine by assigning the "Economy" property to them. These tracks will then use the play engine's buffer (VIP buffer size).

L

Latency

Audio latency and response latency are important aspects of working with audio applications.

Audio latency is the delay of audio data when being processed through an audio function unit (effect plug-ins, DAW audio engines, digital mixers, hardware effect devices, DA/DA converters, etc.). The entire audio latency is made up of the sum of the latency of the sound card (ASIO latency) and the latency of the effects used in monitoring tracks and the master. When monitoring, these latencies are disruptive even at 5 ms.

The response latency is the delay between the use of the sound-creating or sound-changing control element and the audible change or generation of the sound. The response latency is the sum of the output latency from the sound card, the latency from the applied effects, the latency of the play engine and, in some cases, from the MIDI input latency as well as the latency of the graphical user interface of the operating system.

When using MIDI controlled synthesizers (VSTis) the response latency just as critical as the audio latency for monitoring (disruptive after 5 ms).

When playing files with the play engine, response latencies created by effects and the play engine itself are much less problematic (disruptive after approx. 250 ms).

The internal effects show the latency which causes the effect in the lower control element list of the effects. The latency of the sound card in conjunction with the driver being used is displayed in the System Options / Audio Setup dialog (Keyboard shortcut "Y"). The entire latency of all effects is shown in the status bar once playback begins.

Latency comparison

When using effects which cause latency, you must make sure that there is no time delay with parallel tracks, objects, or AUX channels. Depending on the location where effects which cause latency are integrated, various latency comparison methods may be applied. When using latency-causing effects in live input tracks (monitoring), all other tracks must be compensated.

When using effects which cause latency in tracks that are playing content on the hard disk, latency may be balanced using a read offset. The other tracks do not have to be delayed in this case.

When using internal effects in the object and in the master under MME, a real-time latency balance method (intelligent audio stream preview management) is applied.

The advantages of this process:

- When linking the latency-filled effects or increasing latencies by modifying the latency-relevant parameters, passages containing silence will not be created.
- When unlinking the latency-filled effects or the reduction of latencies by modifying the latency-relevant parameters, errors/skipping will not occur.
- Synchronization always remains the same (real-time latency balance – playback not necessary)
- Time changing parameters (timestretching or resampling) may also be used without causing any synchronization problems.

A delay of other audio streams is not necessary.

Lock

This button sets a lock on individual objects or all objects of a track.

If the lock button is active, then individual objects or all objects on a track may be locked in the track header.

Other parameters of locked objects depend on the "Lock definitions (view page 625)" option in the "Object" menu.

M

Markers

Markers act as reminders for position points. They are visible as named orange bars along a special line at the top edge of a project. Markers may be placed during playback as well as during recording.

MIDI object

MIDI objects in Samplitude are edited like wave objects.

Object-oriented editing for audio allows a VIP object to be used for playback instructions of audio files. MIDI objects are structured according to this concept: they refer to a recorded or imported MIDI file and may be copied, split, and trimmed. They have fade handles as well as a middle volume handle.

MIDI objects may also be edited with real-time effects in the object editor. Samplitude displays a different type of object editor specially adapted to MIDI objects for this purpose.

Monitoring

The signal to be recorded is played back through the output device. Different modes are available depending on the driver model used and the monitoring mode. In a strict sense monitoring has nothing to do with the playback of the recorded signal. It really only deals with the signal at the input.

0

Object editor

The object editor enables you to edit the properties of any audio object exactly. For example, the length of the object may be entered numerically or the fade characteristics may be determined here.

Real-time effects, EQ, and dynamics editing may be applied to the object without having to access track effects in the mixer. The selected processing does not depend on any following changes to the object position.

Object-oriented

Object-oriented work describes a method that makes it possible to apply various changes to the audio material connected with the object virtually, i.e. without affecting the actual audio material (data on the hard drive). Samplitude features an independent audio engine behind every object, which enables virtual editing of all DSP functions on the object level.

Objects

In virtual projects, the audio data is represented by rectangles (referred to as "objects") on multiple tracks. An object is a representation of the sample or a segment marked within the sample.

In addition, each object has certain attributes which may be modified using Samplitude's object editor. The rectangle features several handles which may be used to modify the object's start time, length, and volume attributes.

Objects that overlap in one VIP track may be faded into one another. For exact crossfade settings, try the crossfade editor.

The lock symbol positioned on the lower edge of each object may be used to protect a single object against accidental modification. The attributes to be protected can be selected in "Lock definitions".

P

Play cursor

The play cursor or the position line is the vertical line that moves during playback to display the current position in the audio material.

The starting position of the play cursor is moved by clicking in the project. The mouse must be in "Universal" or "Range" mode. If a range is opened, it will be cleared by this action.

R

Range

A range can be stretched or selected in projects using the mouse. As soon as a range is selected, pressing the spacebar will play it back. A range is defined by its start and end (horizontal) as well as by its upper and lower edges (vertical).

The range specifies sections where certain operations are executed (e.g. cutting, inserting, normalizing, fade in/out, move up, calculate effects, and delete).

The range is also used to define loops that are included during project playback.

Routing

Internal rerouting or assignment of signals.

S

Scroll bars

The scroll bars are displayed on the lower and right edges of the arranger window. These bars may be used to navigate the project or to enlarge or reduce the window section.

Scrub

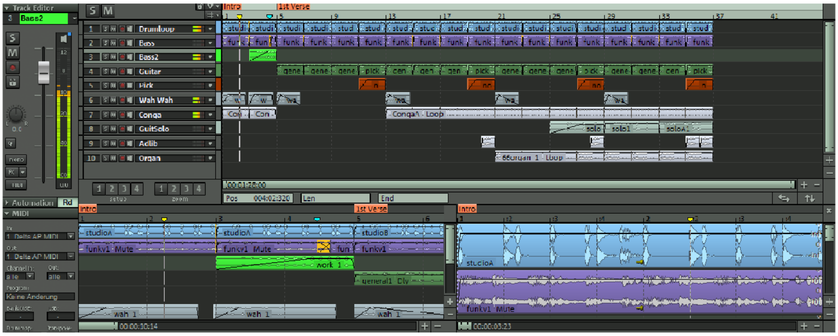
Scrubbing refers to "searching" through the audio material at different speeds to locate a certain position "acoustically".

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

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

Section

"Section" refers to the visible part of a project in the project window. Which section of the project this is depends on the position of the section and the zoom.



There are many commands for moving (scrolling) the visible clip and adjusting its size (zooming). These commands may be opened via the "View" menu, via the position bar, and via the shortcut keys.

Up to three different sections can be displayed in a project at any one time. This way, the entire project can be displayed in one section, while a smaller part where you are working is shown in a different section.

Status Display

The status bar is located at the bottom part of the Samplitude window. For longer actions or calculations a bar is displayed whose width shows the current state of operation.

Action instructions and explanations also appear.

The status bar can also be hidden by going to "View > Toolbars > Status Display" and removing the checkmark.

Submix bus

A submix bus is a track featuring all the abilities of a normal track, i.e. it can include track effects, and panorama and level can be automated. It has one input and one output. The important difference is that you can route the output signal of every track with a lower track number than the submix bus to this bus (instead of routing to the master or an output device).

AUX buses are typically used for effects such as reverb or echo, which may be equally required for various tracks, but in different amounts. In Samplitude, objects may be routed to AUX buses as well.

Surround bus

A Surround bus is required if Surround effects are to be applied. Tracks that include Surround effects therefore have to be routed to a Surround bus where the appropriate parameters may be set.

All tracks routed to a Surround bus have the Surround Editor instead of the normal panorama fader for setting the output level of the track. As soon as a Surround bus is available, the output signal of each individual object may also be routed to this bus and positioned in the Surround panorama independent of the track panorama settings.

From a technical viewpoint, the Surround master is also a Surround bus in a Surround project, since Surround channels may also be assigned to the audio devices and it cannot be routed further to a Surround bus.

System settings

Settings that are not project-specific are described as system settings. These include settings that apply to the entire project.

T

Tempo markers

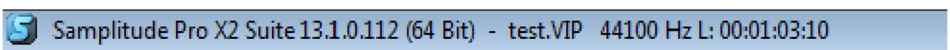
A tempo marker indicates a tempo change at a certain project position.

Time input field

All fields that represent a time position can act as an input field by double clicking with the mouse. The time units may be changed by clicking on the right section of the field.

Title bar

The title bar is located at the top of the window. It contains the name of the application and the project.



To reposition the window, simply click the title bar and move it as desired. You can also move around dialog boxes by moving the title bar.

Tooltips

Tooltips provide additional information about the function of a certain control element. Hold the mouse pointer over an element for a few seconds, and more information will be displayed. Text featuring the name or a short description of the element will appear. Keyboard shortcuts and operating instructions are often also included.

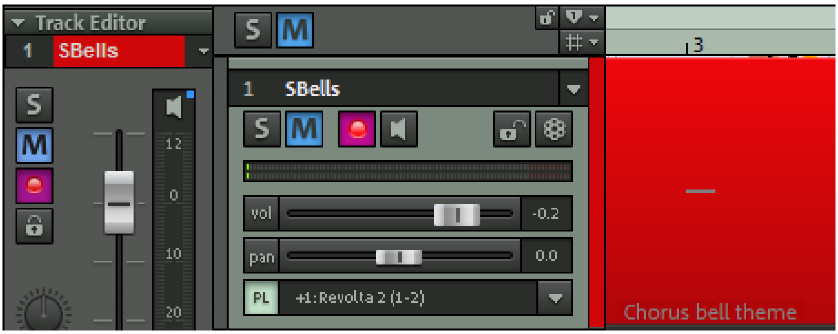
Track

The audio track. A track can contain audio and MIDI objects. Automation curves are also represented in the track.

Track Header

The track header is the front section of a VIP track. It contains various functions including mixer elements, buttons for loading plug-ins, and other track parameters.

Track header with activated mute



You can show or hide the track header using the "Tab" key. The display options can be opened with "Shift + Tab". The "Buttons/Slider" option in the "VIP" section ensures that the track header is displayed as the default standard.

V

VIP - Virtual Projects

Virtual projects (VIP) are the top level of audio editing in Samplitude. In virtual projects, objects can arranged to create complex arrangements. All editing steps carried out in the arranger window are applied non-destructively. This allows you to select the proper cut position, cut length, volume level, effect setting, etc. without losing the original audio material or altering it.

The actual audio data is represented by objects on the various tracks. An object is a representation of the sample or a segment marked within the range. In other words, the object references the audio file.

Visualization

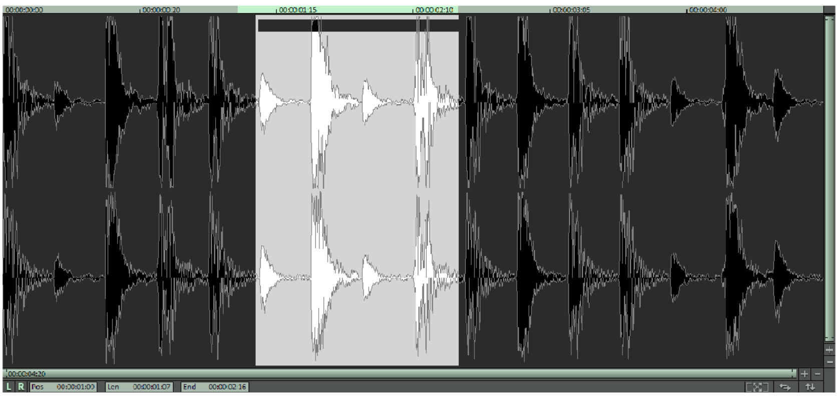
The visualization feature displays the audio material visually in different ways.

W

Wave editing

Non-destructive wave editing lets you take advantage of combining direct and virtual editing of audio material (wave projects), just like with VIP projects.

Wave projects



A wave project contains audio data. Objects in the VIP refer to this audio data.

If you use Samplitude as recording/mastering DAW you usually don't have to edit wave projects as all editing options can be performed quicker and safer in the virtual project. For using Samplitude as a wave editor, refer to "Working in the project window -> Samplitude as a wave editor (view page 66)".

The windows of the wave projects used by VIP objects are usually hidden. These may be accessed via "Edit wave projects" in the "Object" menu or by pressing "Shift" and double clicking an object used by wave project.

All loaded wave projects may be made visible as icons via the command "View > Window > Iconise All Wave Projects". To hide them again, go to "View > Window > Hide All Wave Projects".

The name of the wave project, sample length, and the resulting memory requirement may be entered in the title bar of the wave window. To activate a wave project you have to click into the corresponding window. Samplitude can manage an infinite number of wave projects on the screen.

Work area

Work areas are for adapting the program to certain work styles, i.e. they may be changed, renamed, and created. The work area is opened via the "Workspace bar". Work areas are available as wave editing or virtual projects.

Menu Reference

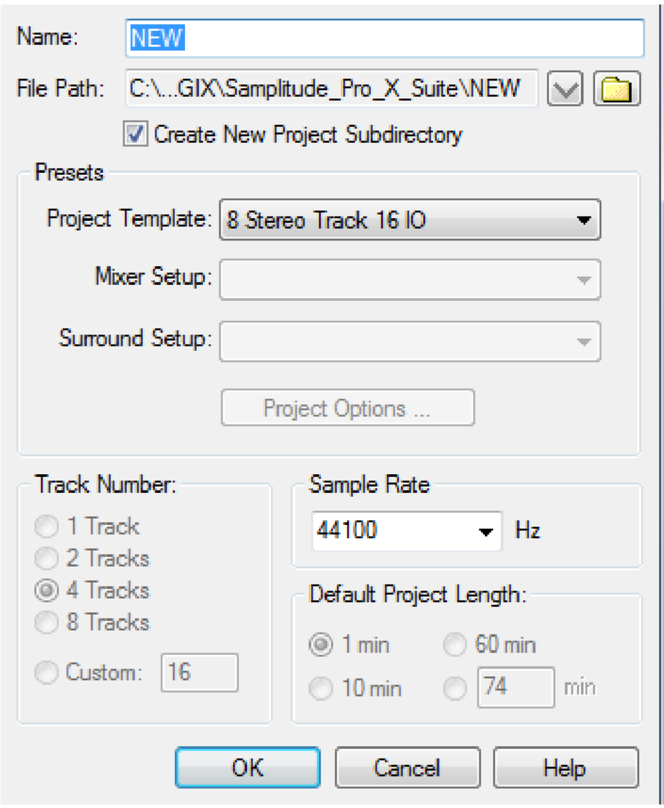
Extensive information on the individual menu points of Samplitude can be found in the following sections.

For a better overview, not used menu items can be hidden. If menu items are described in the following sections that are not present in your menus, they are probably just hidden. To display them again, activate them in the menu settings by going to "File -> Program settings > Edit keyboard shortcuts and menus (view page 626)".

File Menu

This allows you to apply comprehensive settings for creating, managing, loading, and saving projects.

New Virtual Project (VIP)



Shortcut: E

A new Virtual Project is created in the arranger with the specified number of tracks:

Name: Enter the name for the new project.

File path: Select the folder for saving the project.

Create new project subdirectory: A subdirectory featuring the name of the project is automatically created in the directory selected in "File path".

Project template: Choose between previously selected project templates in this list field. These presets contain project settings such as track number, device assignments, etc. Create project templates with "File -> Save project as template (view page 574)"

Mixer setup: Select a preset mixer setup here. Mixer setups define the number and kind of tracks/buses as well as the routing of inputs/outputs and effects.

To create a Surround project, define the Surround setup first. The list field located below, i.e. "**Surround setup**", is activated for selecting a number of Surround formats.

Project options: All settings adjustments made here are valid for every newly-created virtual project. They will be saved in a separate VIP(Templates\Template.VIP). The Template. VIP file may also be edited directly to define additional settings for all projects:

- Record arming setting of the first track
- Some mixer setup settings
- Playback mode
- Effect sequence on the tracks and mixer
- Grid offsets

> Please also read "System settings -> Project options -> General (view page 84)" for more details about project options.

Track number: The start track count of the virtual project is defined here. Use "Track -> Insert new tracks... (view page 669)" to add tracks at any time.

Sample rate: Defines the sample rate of the virtual project.

Preset project length: The initial project length may be set as a preset here. This setting will be automatically adjusted during loading or recording and changes during zoom operations.

Note: VIP projects may only be up to 168 hours in length.

There are no more length restrictions on WAV files with NTSF. Recordings are written to a large RIFF64 file. If under 2 GB, it will be compatible with normal wave files. FAT 32 continues to split these into W01, W02, etc..

Open

You can open various file types here and load them into Samplitude. To load audio files, use the command "Import -> Load Audio File".

Virtual Project (*.VIP)

Use this menu command to load a multitrack project as an arrangement.

Shortcut: O

RAM Wave (*.RAP)

RAM Wave Projects contain audio data in an audio format unique to Samplitude that is loaded into the RAM memory. Additional information for display of the audio data, marker positions, etc. is contained by this data.

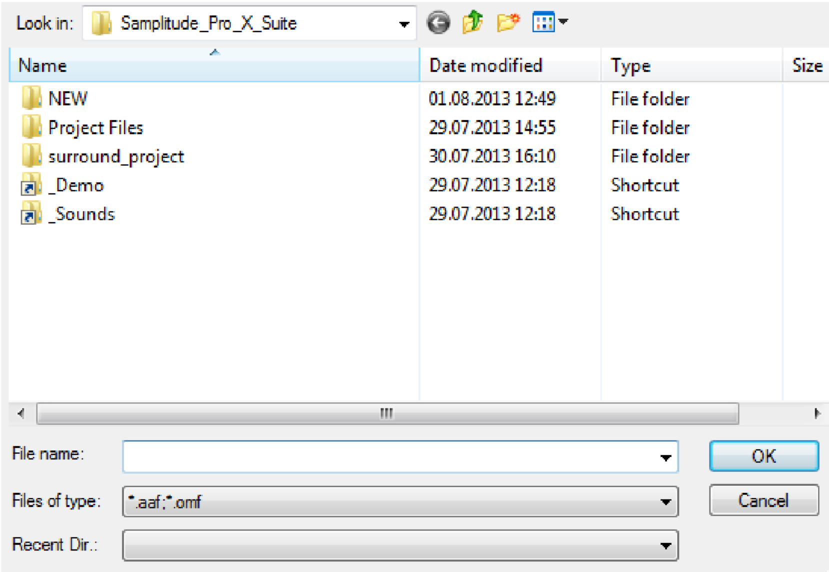
HD Wave (*.HDP)

HD Wave Projects contain audio files that are loaded directly from the hard disk. Additional information for display of the audio data, marker positions, etc. is contained in them. The files use WAV audio format.

Shortcut: Shift + L

Note: If a VIP is active, the loaded wave projects are loaded immediately into the VIP as objects. The range of the active track displays the position. "CD arrangement" mode is the exception. In this case, the objects are displayed below one another independent of the range. The distance between the objects corresponds to the pause time (CD/DVD menu -> Set start pause time (view page 1008)).

Import AAF/OMF



The**AAF import** transfers the following content:

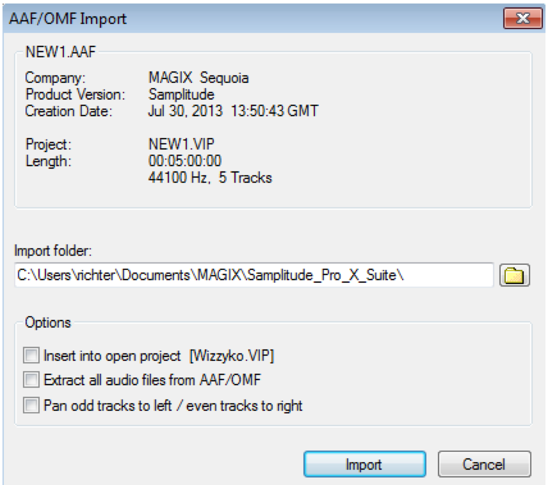
- Object position/wave offset - sample exact
- Object fade in/out - linear only
- Object crossfade - linear only
- Object volume
- Object pan
- Object volume automation
- Object pan automation
- Track names
- Track Volume
- Track Pan
- Track Volume Automation
- Track Pan Automation
- VIP markers
- Timecode offset
- Timecode format (24, 25, 30 fps)
- Processing timestamps in audio files

The**OMF import** transfers the following content:

- Object position/wave offset - sample exact
- Object fade in/out - linear only
- Object crossfade - linear only
- Object volume
- Track names

Note: When importing AAF/OMF files video files will also be imported. These can only be played when the program 'understands' the format and when the corresponding codecs are available on the system.

Find the AAF/OMF file you want to import using the import dialog. In a further dialog you can choose the track panorama when importing, so that the uneven track numbers in the panorama are placed on the left and the even on the right.



Object (*.OBJ)

An object contain playback instructions for the underlying wave project. The object contains information about the time position as well as the editing steps involved. Objects are used in virtual projects.

Edit List (*.EDL)

The EDL helps exchange projects with other sequencer programs. A virtual project that resembles the EDL is created. The list is a text file that contains information about the wave files used, video files, object borders, object volumes, markers, and volume & panorama automation.

Table of Content (*.TOC)

With the table of contents (toc) of an audio CD the information about the contents of a CD is saved.

Session (*.SAM)

This command loads a Samplitude session. This opens all projects and arranges the windows as they were distributed on the screen when the session was saved.

Import

Load audio file

Here you can open a dialog for loading of audio files.

The following formats are supported and read directly by Samplitude:

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

Note: Open all other formats with the import audio function "Import audio" to convert these to wave files for local storage on the hard drive.

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

If a virtual project is opened, new objects are created simultaneously in the VIP which refer to loaded audio files. If an area has been selected in the VIP, the files are positioned at the beginning of the range or with a separating pause of 2 seconds after the last object. The length of the pause can be adjusted in the "CD/DVD" menu via "Set pause index".

If you want to load the audio file as a separate wave project, click the corresponding check box.

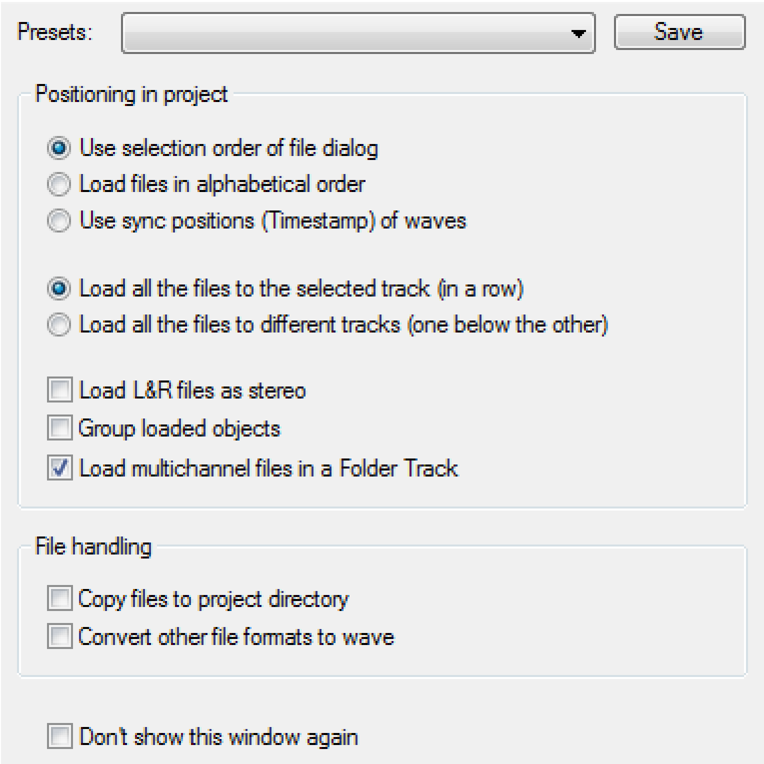
Clicking the payback button enables each audio file to be listened to in advance.

Attention: The monitoring function uses the Windows Multimedia MME standard output device. Many audio cards shut down the Windows MME system when ASIO drivers are used, which means problems may occur during monitoring. For this reason, the preview function is deactivated by default when ASIO drivers are used.

The "load audio file" option activates the monitoring function again at any time.

The "options" button opens the options for loading an audio file (view page 569).

Options for loading an audio file



Use selection order of file dialog: If this option is activated, Samplitude remembers the sequence in which the files were selected and then sorts them accordingly.

Load files in alphabetical order: If this option is activated, Samplitude sorts the selected files alphabetically in the VIP.

Load files to sync positions (timestamp in wave): Broadcast wave files containing time stamps are positioned precisely at this position in the VIP.

Load all files to selected track: The selected files are loaded successively in one of the selected tracks.

Load files to different tracks: The files are now sorted in vertical order from the selected track to the next one. If necessary, an additional track is added.

Load L&R files as stereo: Here you can load files separated by left and right stereo channels as stereo files.

Group loaded objects: All loaded files are grouped. They can be ungrouped anytime.

Load multichannel files into a folder track: This option is activated by default. When loading multi-channel files (i.e. files with more than two channels), it is assumed that these are Surround files in Interleaved format. A folder (view page 136) track with the required number of tracks is created and routed to the surround master (that is also created if necessary). The tracks in the folder track get the surround panning according to the determined surround format. If this option is not active, the individual channels are loaded onto the track and the tracks below it.

Other (only present for "Load audio file")

File preview with Windows Media Player Control: Files may be previewed via the Windows Media Player.

File Management

Copy audio files to project directory: Copies the file to the corresponding project folder.

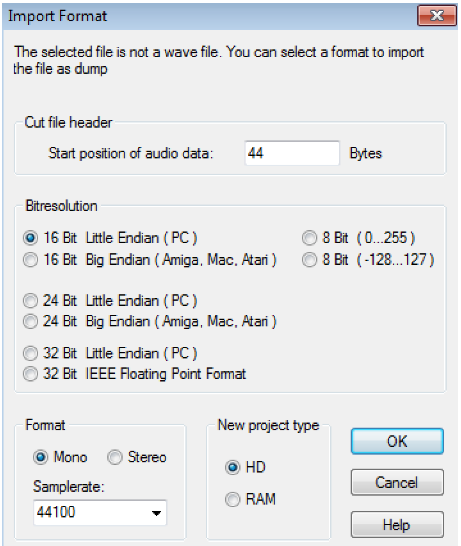
Convert compressed files to wave format: Compressed audio formats such as MP3 can also be loaded and played directly in Samplitude. This requires increased CPU calculation. Activate this option to convert files like this into an uncompressed wave format.

Sound Designer II

Use this function to load Sound Designer II files.

Import raw data (dump)

With the "**Import raw data (dump)**" you can try to import damaged files that sound like white noise on playback. The header is usually damaged or missing. In this case you can enter "0 bytes" as the header length in the following dialog.



Wave or MP3 projects will be imported as raw audio data (PCM) in RAW format (Little Endian for PC, Big Endian for Amiga, Mac, Atari).

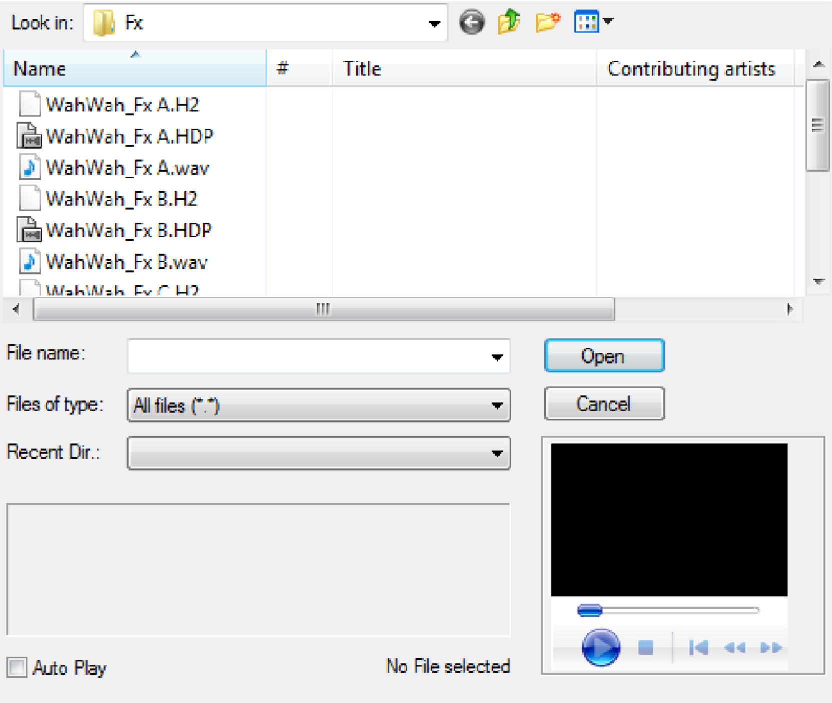
Load MIDI File

This function loads a MIDI file. Detailed information about this topic is provided in "MIDI in Samplitude -> MIDI - record, import, edit -> Importing MIDI files (view page 324)".

Keyboard shortcut: Shift + M

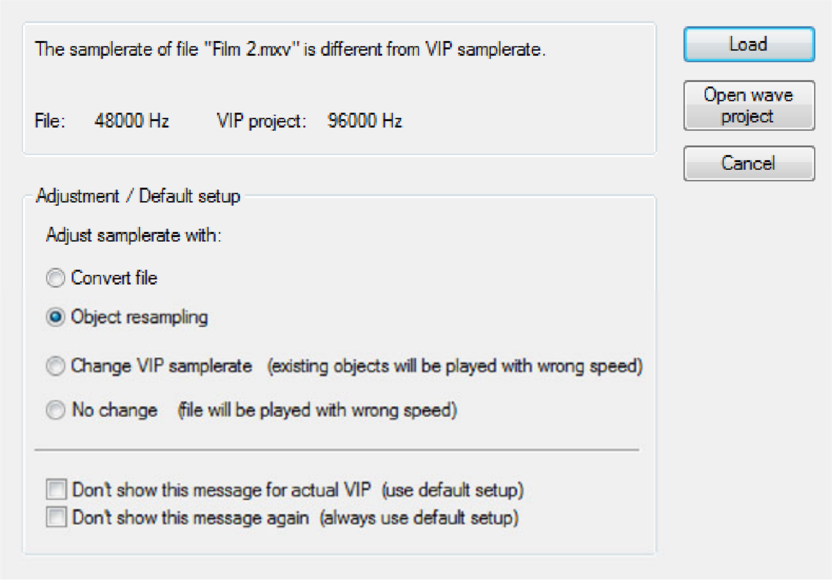
Load Video File

This dialog helps load a video into the current project. After selecting a video, preview it or inspect its properties. If a check is placed next to "**auto play**", the selected video file will play automatically in the preview window.



Before loading the video, you will be asked if you want to load the video's audio track into the project. This is positioned in the track underneath the video track. If the sample rate of the audio file deviates from the VIP's sample rate, then the sample rate may be adjusted. The following options are available for adjusting the sample rate:

- Convert file: Samplitude creates a converted audio file in the VIP's sample rate. The file name will now also indicate the new sample rate.
- Object resampling: If this option is active, Samplitude will resample the object. The quality of object resampling may be set under "**System option -> Effects -> Resampling/bouncing** (view page 615)".
- Adjust VIP sample rate (existing objects are played back at incorrect speed)
- No change (file plays back at incorrect speed)



Detailed information about displaying video objects is available in "Video setup".

Import Audio CD Track(s)

This command imports the audio contents from a CD into Samplitude (view page 982).

Import Audio DVD

This command imports the audio contents of a DVD into Samplitude (view page 985).

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

Save Project

The current disc project is saved with the name displayed in the project window. If you have not yet specified a name for your project, Samplitude will ask you to do so.

Shortcut: Ctrl + S

Save Project as...

A standard dialog will open, where you can specify the path and name of the project to be saved. If you would like to save wave projects and virtual projects under a new name and in a different location, the original file will remain untouched. Then you can continue working in the new project (automatic).

Shortcut: Shift + S

Save Project Copy...

A dialog open to create a copy of the project, which may be saved under a different name. In this case, the date and time are added automatically to the project name. Of course, you can name the copy any way you like. Once saved you will be able to continue working on your original project.

Save Project as Template...

This option saves the current project as a project template. The template will save all project-based settings like grid form, track number, etc., but not the used objects or wave files. Templates may be loaded when creating new multitrack projects ("File" > "New Virtual Project").

Save project as EDL...

The active virtual project will be saved as a list in the EDL text format. This edit list is a text file containing information about used wave files, video files, object borders, object volumes, markers, volumes & panorama curves.

This option effectively lets you save Samplitude projects into an exchange format that allows a project to be incorporated into a different sequencer or video editing program. EDLs may be converted together with the audio files into other formats using a conversion program like EDL ConvertPro.

Save Object...

This function allows you to save a single object. This file contains all the data of the object editor like fades, effects, object names, etc.), but not the actual audio data. The wave project referenced by the object also has to be available if you want to reload the object.

Save Session...

A Samplitude session is a meta project that includes the file paths and window positions of all loaded projects upon saving. This is useful for starting to work again at the same position where you ended, without having to load various projects.

If you save a session with the name "startup.sam" and place it into the directory with the program folder, Samplitude will load that session automatically the next time you start the program.

Export

This function has been revised in Samplitude. For the latest information, see the PDF document **Samplitude Pro X7 New Features** in the program folder.

This menu option allows you to export projects in various formats. The export dialogs for the respective formats are almost identical.

File name and path selection: In the upper part of the dialog you'll find a file explorer which you can use to set a new name or target folder for the audio file. With "Recent Dir." you can choose from a list of save locations that are already being used.

File type: Here you can set the format for the exported file. Format settings: Here you'll find the settings and options for the corresponding export format. (see below)

Export settings: Under Export Settings you will find a menu where you can make further settings for the export. Selecting a menu item enables or disables the corresponding option.

In the upper part of the menu you can specify which part of the project should be exported:

- **Marked Range only:** Here the export will be conducted only over the length of the range selected in the arranger.
- **Complete project (up to last object):** The time selection for export goes from the beginning of the project to the end of the last object plus the reverberation time. This is the default option.
- **Complete CD:** If you select this option, the project will be used until a CD end index is created, without such an index the whole project will be used until the end of the project.

The next section of the menu determines whether the export is to a single file or to several separate files whose beginning is determined by track markers or other markers:

- **Create one file:** A single file is created.
- **Each CD track in a file:** Individual files are created, ranging from one CD title index to the next.
- **Each CD track in a file (from track to pause marker):** Individual files are created, ranging from one CD title index to the next CD pause index. The sections between this pause index and the next CD track index are not exported.
- **Any marker:** Individual files are created, each of which extends from one marker to the next.

Note: When exporting to single files, the name of the corresponding marker is used as file name instead of the specified file name.

More options:

- **Use track number as prefix:** When exporting to individual files, the file name (marker name) is preceded by a consecutive number.
- **Export CD markers only:** If supported by the export format, all markers present in the project are saved as audio markers in the exported audio file. With this option you can limit this to the CD track markers.
- **Write cue list:** When exporting to a single file, it is possible to save contained markers in a cue list of the same name as a .cue file.
- **Write playlist file:** When exporting to multiple files, a playlist of the files is saved as a *.m3u file.

WAV

Here you can export Wave files.

Format settings: A dialog window featuring a list of all compression codecs appears here. Select the codec and compression rate you require. If you click the "**Dithering**" button, the dithering options dialog will open. Detailed information about this is available in "File > Program Preferences > Dithering Options (view page 638)".

MP3

This exports the project as an MPEG Layer 3 with the included encoder. Clicking on format settings opens a window for setting encoder options. Here you can set the export format, encoder quality, stereo coding, VBR (variable bit rate) options and labeling using the MP3 ID editor.

MPEG...

Exports the project as an MPEG file. These formats may also be set in the advanced settings dialog, which is opened via the button "Format settings". The options "stereo", "joint stereo", and "mono" are featured.

Windows Media...

Exports the project in Window Media Format.

This format is an optimized audio/video format for the Internet from Microsoft. This format also enables streaming playback of audio files over the Internet. Select a profile for the Windows Media file in the "format settings" dialog. Enter the track and artist names, copyright details, and descriptions.

AAC

Exports the project as an AAC file (Low Complexity or High Efficiency). The encoder can be set up in the advanced dialog that is opened through "Format Settings". You can also set the quality and transport format here. AAC export uses the file ending *.m4a.

> **Note:** During AAC export the sample rate varies depending on the bit rate.
>
> For example, if you load a 44.1 kHz wave file into a 44.1 kHz VIP and then export it as an AAC with 128 kBit/s, you will get a 44.1 kHz file.
>
> However, if you load a 44.1 kHz wave file into a 44.1 kHz VIP and export it as an AAC with 256 kBit/s, you will get a 48 kHz file.

FLAC...

The FLAC format allows you to compress 16-bit or 24-bit files during export. The properties of the FLAC encoder may be accessed via the "Format settings" button. Set the compression degree, bit resolution, format, sample rate, and dithering here.

OGG Vorbis...

Exports the project in OGG Vorbis format.

The "Format settings" button opens an additional dialog to select the desired bit rate. The scale of compression ranges from 46 kBit/s to 500 kBit/s (for MP3 from 32 kBit/s to 320 kBit/s). In the stereo options, choose between "Stereo" and "Mono" and also select the "Variable bit rate" mode.

AIFF...

The project will be exported in AIFF format. In the advanced dialog, select the bit resolution, stereo format, sample rate, and quality level of sample rate adjustments.

Mono / Stereo Convert

Stereo Wave -> 2 Mono

This function saves both the left and right channels of a stereo wave file (file.wav) separately as a mono file (file_left.wav, file_right.wav).

"Stereo to 2 Mono" may also be applied to objects in the VIP. Stereo objects to be split are replaced by two mono objects that are arranged one below the other and which feature panorama settings at the object level to the left or right.

2 Mono -> Stereo Wave

This combines two mono wave projects to one stereo wave project.

After executing this command, a dialog opens displaying all loaded mono files. You can load additional files with "Load file". From the opened wave projects, select the right and left files with the "ˆ" keys. With "<->" keys you can switch channels. After clicking "Link files", specify the name of the new file to be created in the project folder.

Please note that only mono wave projects of the same word width and sample rate can be linked. The length of both projects are adapted to one another, the longer mono file sets the total length.

LR Wave -> 2 Mono

LR wave projects may be split into two independent mono projects using this function.

LR Wave -> 1 Mono

The active LR wave project is set to mono mode. In this case, the two channels are mixed together by adding the corresponding samples each at 100% and then splitting the sum in two in order to avoid overmodulation. This corresponds to sinking the level by 6 dB.

2 Mono -> LR Wave

This creates an LR wave project with two channels from two mono wave projects, in which case one mono wave project in panorama view is positioned to the left, and the other one to the right. Select both mono files with consecutive mouse clicks. In the following dialog you can decide to "Link". "Convert to stereo" creates a stereo wave file.

1 Mono -> LR Wave

In this case, LR wave project with two identical channels is created from a mono wave project. The original sample is duplicated here.

Save in format

Use this function to save wave projects in various formats.

This can be useful when, for example, RAM projects are to be converted into HD Wave Projects or LR Wave Projects (two mono samples linked to one another) are to be converted into Stereo Wave Projects.

In "**Area settings**", select whether only the **selected area** or the **entire wave project** should be saved in the new format.

Under "**General options**", select whether the **maximum amplitude** should be displayed or not and if the **master effects** should also be copied when the file is saved.

The **Dithering Options** (view page 638) can also be adjusted here.

In the "**Format**" section you can determine in what format the file should be saved (**Wave**,**MP3**, **AIFF**, **Ogg Vorbis** or**FLAC**).

In the case of Wave, AIFF and FLAC projects you **also** have the **options "Stereo"**, **"left & right"**, **"only left channel"**, **"only right channel"**, **"mono mixdown"** and **"Mono"** and the **selection/quality of the sample rate** options open to you.

By clicking on the **format settings** for **MP3 format** a new window opens, where **encoder options** can be defined. Here you can set the **Output format**, **Encoder quality**, **Padding-mode**, **Content options**, **VBR** (variable bit rate)**options** and tagging using the **MP3 ID Editor**.

In the dialog for **AIFF** you can specify the **bit resolution**, **stereo format**, **sample rate**, and **quality level of sample rate adjustments**.

Change Bit Resolution

Select the desired word width/bit resolution of wave projects here.

Working with the 32 Bit Float Format

If you convert fixed comma wave projects into the floating comma format, the signal will remain a 16 or 24-bit signal and the quality will not be improved. This will only yield a result if you are applying destructive changes to the audio material as precise calculation is practically impossible after editing.

You use overmodulation resistance and level-independent dynamics. The quantization hiss will not increase, even if you set very low levels.

The downside is that twice the amount of hard disk space is required and only half the tracks are now playable simultaneously (depending on the CPU).

If you convert from 32-bit float to 16 or 24 bits, dithering occurs to reduce subjective quality loss.

Working with 8 Bit Wave Projects

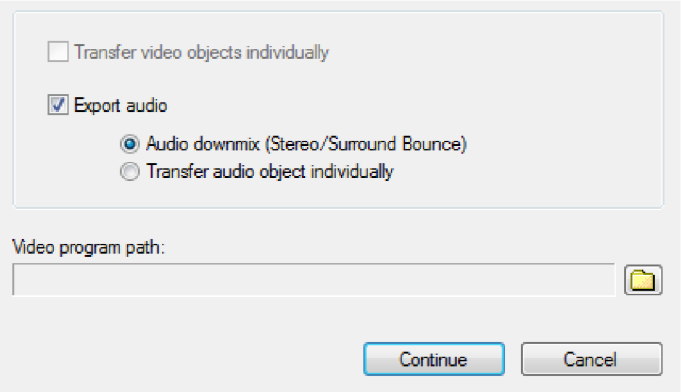
A smaller word width may be useful when editing audio for multimedia applications, since these applications may use a resolution of 8 bits to save storage space.

The downside of such a low bit resolution is that as the quality of the so-called SNR (signal-to-noise ration) drops, and the quantization noise increases. This quantization noise is not consistent, but is modulated by the signal and may be unpleasant.

The word width of the project is always indicated in the title bar of wave projects. To perform manifold destructive editing on an 8-bit project, you should convert it into 16 or 32-bit float before editing. The created calculation precision will then occur in the 16-bit range, this being considerably smaller. After editing, convert the sample back to 8 bits.

Export to "MAGIX Video Pro X"...

The following trackbouncing dialog opens for exporting to the external MAGIX program "Video Pro X".



Select whether only the video objects should be transferred, whether an **Audio downmix (stereo/Surround bounce)** should be made, or if the **Audio objects should be transferred individually**. If the "Continue" button is pressed, a save dialog will appear.

Export Video Sound...

After successfully editing the video sound in Samplitude, you can write the sound back into the video file. You may replace the original sound of the video or create a new video file.

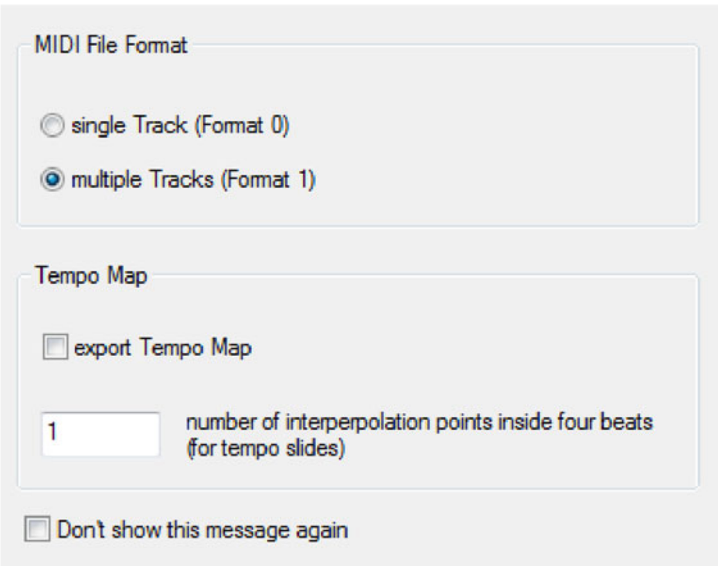
Set the source file under "Video source".

Replace audio in the original AVI file: This option carries out trackbouncing on the current VIP. This creates a temporary wave file which can then be integrated into the selected AVI file. It's necessary that the AVI contains an audio track of the same format (sample rate and bit resolution).

Save a new AVI file to: This option applies trackbouncing to the current VIP. This creates a temporary wave file which may be copied with the selected AVI file into a new AVI file. This process contains the original AVI file.

Note: If the lengths of the audio and video files differ from one another, a warning dialog opens. After the export, the longer component is cut off; an attempt to automatically synchronize the data is not carried out. If you receive this warning, try to resample the soundtrack to the correct length.

Export MIDI File...



The particular file format may also be specified when exporting a MIDI file. One track **corresponds to format 0**, **multiple tracks to format 1**. In addition, the tempo map may also be exported, in which case the support point count per beat for tempo runs can be selected freely.

The standard MIDI file (SMF) export always takes place in the current VIP-PPQ resolution. The markers are also exported.

Exporting MIDI Objects as Individual MIDI Files

This command exports all selected MIDI objects as separate MIDI files.

List Export

A current **marker list**, **object list**, or **track list** for your project can be written as a text/CSV file and stored in the project folder. You can select the parameters that should be included in the list in the export options.

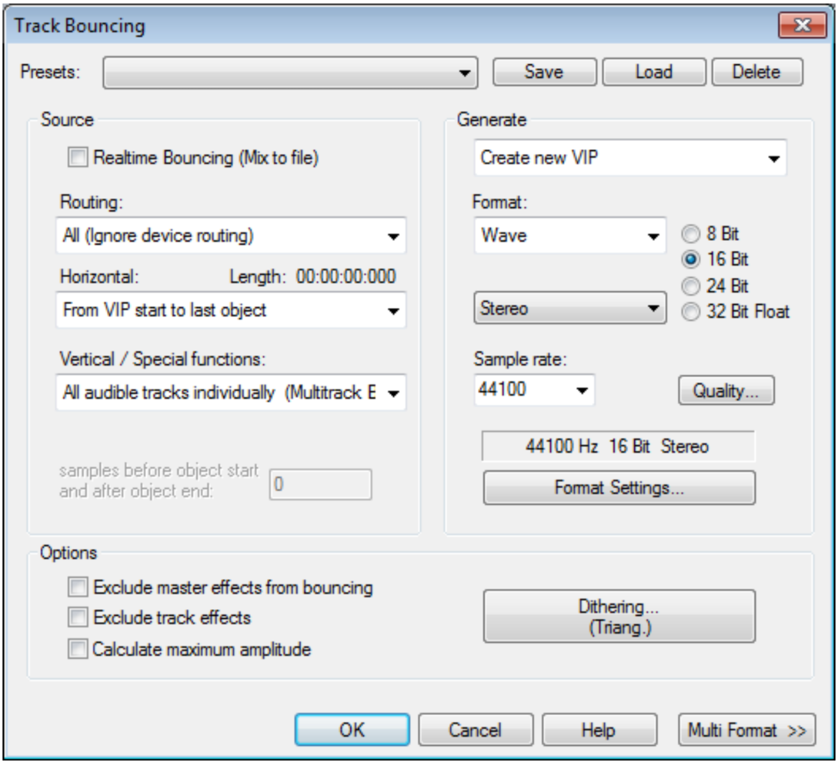
Advanced export & Trackbouncing

This function has been revised in Samplitude. For the latest information, see the PDF document **Samplitude Pro X7 New Features** in the program folder.

For exports that go beyond simply rendering the project into an audio file, use the Track Bouncing dialog. Here you can create one or more audio files by combining the project or a section of the project, selected objects and tracks.

There are other differences to the simple export:

- Note: All files created as a result of trackbouncing are opened in Samplitude. This means that you cannot bounce into the same audio file twice in a row, but have to close it first. If you bounce often and do not require any special mastering options, you can work more effectively by using the simpler "File > Export" (view page 575) command instead of the track bouncing dialog.
- All formats that cannot be directly opened by Samplitude cannot be created in the track bouncing dialog. For bouncing into AAC, MPEG or WMA formats, use the "File -> Export (view page 575)" command.
- The export settings related to CD track markers, e.g. to create single files at CD track markers when exporting, are not available in the track bouncing dialog.



All settings made in the dialog can be saved as presets.

Source

Under **Source** you define which audio material from the project should be used for track bouncing.

realtime bouncing (Mix to File) The "Mix to File" (view page 218) function allows you to quickly perform a mixdown in real time and change your mix during playback.

Routing: This drop-down menu is used to define the outputs whose signal output should be written to the file(s).

- **Master:** The mixer's master output is written to a file. This is the default setting for exports.

If a surround master is present, there are the additional options

- **Surround master** (view page 320): This bounces the surround mix.
- **Surround + Stereo Master:** The surround mix is bounced together with the stereo mix.

The remaining two options are only relevant if bouncing of individual tracks is selected under "Vertical/Special functions".

- **All (Ignore Device Routing):** A mono or stereo file is created for each unmuted track (see Track Bouncing Settings: Format), regardless of its routing to the outputs.
- **All Devices (Use device routing):** All unmuted tracks are combined according to their routing to the individual outputs. For each physical output, individual mono files, stereo files or an Interleaved RIFF Wave (Multichannel Wave) with the corresponding number of channels are generated, depending on the settings under **Format**.

Horizontal: In this drop-down menu, select the time range of the project for bouncing.

- **Marked Range only:** Here the export will be conducted only over the length of the range selected in the arranger.
- **From VIP start to last object:** The time selection for bouncing goes from the beginning of the project to the end of the last object plus the reverberation time. This is the default option.
- **Complete project:** If you select this option, the complete virtual project will be bounced, including the silence after the last object until the project end.

Vertical/Special functions: Use this drop-down menu to specify whether you want to combine the output of all tracks or only selected tracks into one file, create separate files per track, or bounce individual objects.

- **All audible tracks:** All unmuted tracks are used for bouncing.**This is the default option.**
- **Only selected tracks**: All selected tracks are used for bouncing.

The following two options are only useful if **All (Ignore Device Routing)** is selected under Routing.

- **All audible tracks individually (Multitrack bounce):** All tracks that have not been muted are individually calculated into new files.
- **All selected tracks individually (Multitrack bounce):** All selected tracks that have not been muted are individually calculated into new files.

For the last two options, there is an additional setting **Samples before object start and after object end**, with which you can include additional samples before and after the object boundaries in the file to be calculated.

- **Bounce all selected objects individually:** All selected objects will be bounced individually, including the object effects, into a new file. This enables a large number of single objects to be exported quickly to separate files (e.g. for creating sample archives).
- **Glue selected objects together**: Per track, all selected objects are processed and written into a new file including object effects. In this process, all objects

between the frontmost and the last selected object in each affected track are glued together.

Notes:

- If a **Surround Master** or **All (Use Device Routing)** is selected under **Routing** (see above), track bouncing of individual tracks or objects is not available.
- Note: If a selected track is a submix bus, all tracks that are routed into this bus are also bounced. This is done recursively, i.e. if these tracks are busses again, their sources are calculated as well. This permits fast mixing of individual groups.

Track bouncing settings: Generate

In this drop-down menu you can define what should happen to the generated audio files after track bouncing.

Create new audio file: The audio file or audio files are simply created.

Open new wave project: The audio file or audio files are created and opened as wave projects in new windows.

Create new object in VIP: The audio file or audio files are created and inserted as new objects in a new track in the existing VIP. This selection automatically activates the option "Exclude master effects from bouncing" to prevent master effects from being applied twice.

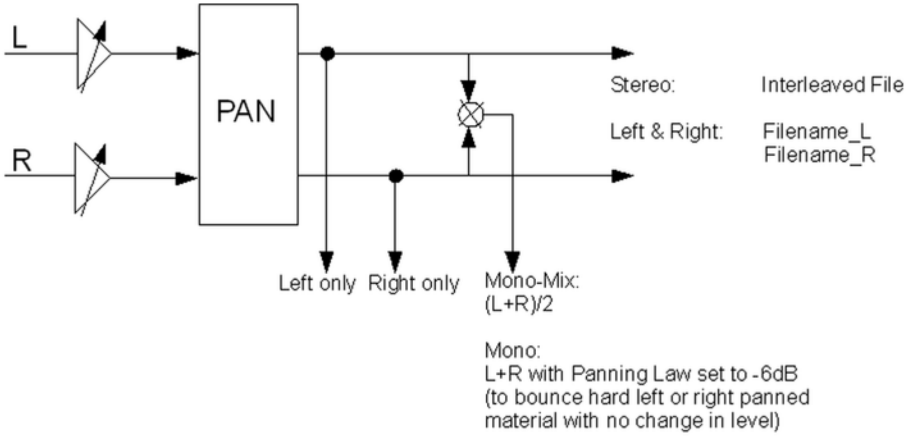
Replace object(s): The objects used in trackbouncing are removed and the result is inserted into the selected track. This selection automatically activates the options (see next section) "Exclude master effects from bouncing" and "Exclude track effects" to prevent these effects from being applied twice.

Create new VIP: A new virtual project (VIP) will open. The bounced audio material is inserted into the first track of the created project.

Trackbouncing criteria: Format

The target format may be set flexibly. **Wave** and **AIFF** files in 8/16/24/32-bit mono/stereo at variable sample rates and four different quality levels are supported, as are **MP3**, **Ogg Vorbis**, and **FLAC** files at **variable sample and bit rates**. The desired codec can also be selected by clicking "**Format settings...**"

The following diagram displays the signal flow for the target format's setting.



- The format setting "**Stereo**" creates a file that contains the stereo information as an "**Interleaved file**".
- The format setting "**Left & right**" can be used to create two files, i.e. "**Filename_L**" and "**Filename_R**", which provide separate information about the left and right channels.
- The format setting "**Left channel only**" monitors and outputs the **left channel only**.
- The format setting "**Right channel only**" monitors and outputs the **right channel only**.
- The format setting "**Mono mixdown**" calculates the left and right channels together according to the formula "**(L+R)/2**" and then outputs these.
- The format setting "**Mono**" is especially useful for hard left or right tracks, since it bounces these at the same level. Mono is formed according to the formula "**L+R**". If **panorama** is set in the middle in the format settings, the trackbouncing process will automatically result in a **level reduction of 6 dB** per channel (panning law -6 dB). This will ensure that even centered mono signals are not bounced at a too high level, but at the original level.

If you set **Surround Bouncing** (view page 320) in "Source -> Routing", the formats "Mono", "Stereo", and "Interleaved file (RIFF 64)" will be available for selection.

Track Bouncing Criteria: Options

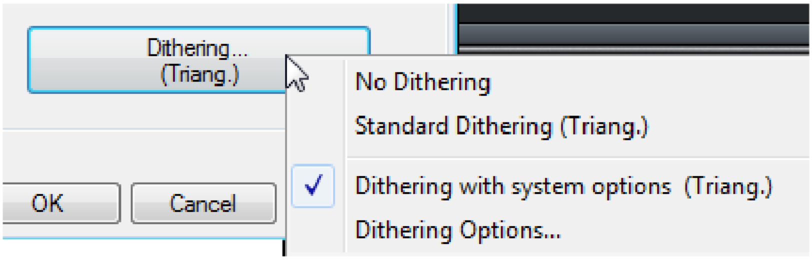
Exclude master effects from bouncing: Master effects are not included during bouncing.

Exclude track effects of target track: Track effects of the target track are not included if they are identical with the source track.

These options are important in combination with the "Generate" functions "Create new object in VIP" and "Replace object(s) in VIP". Note: these are set automatically so that the track effects are not applied twice.

Calculate maximum amplitude: Displays the maximum volume in dB after bouncing in order to correctly set equipment for additional editing or to correct the master level. Once bouncing has been completed, the corresponding information window will be displayed.

Dithering: Every track bouncing process can have its own dithering process, independent of the global settings. This enables dithering to be bypassed or to apply the standard dithering (dithering featuring triangular noise).



Use this dialog to apply dithering according to the system options or to access the dithering options in the system settings. The button value in brackets (e. g. **Triang.** or **POW-r 1**) indicates the currently set dithering algorithm.

Detailed information on this can be found in the menu reference under "File" > "Program Preferences" > "Dithering Options (view page 638)".

Range Bouncing (Internal Mixdown)

Use this function to convert the objects within a selected range to a new wave file. The objects are then replaced in the arranger. This is useful if you want to combine multiple objects into a single object for easy editing later.

To create performance room you can render real-time object effects using the range trackbouncing feature.

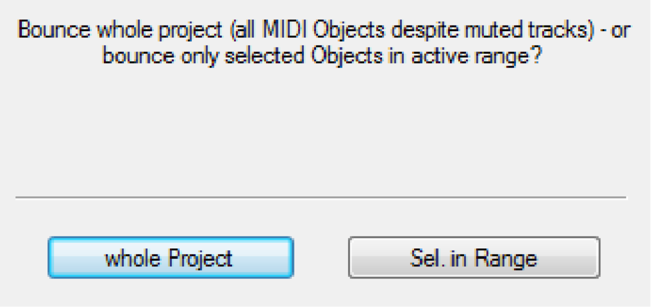
If the rate of your objects is higher than 16-bits, a dialog box will ask you whether the new file will be created as a 32-bit float or 16-bit file:

32-bit (float): The resulting file features a 32-bit (float) rate. This setting is practical if 24-bit objects or float objects are used in the bouncing and their high rates should be retained.

16-bit (integer): The resulting file is 16-bit rate. This setting is practical when the recording needs to be burned to CD and when there are no 24-bit objects in use. If 24-bit objects are being used during bouncing, these may be reduced to 16-bit using dithering options.

MIDI bouncing

In another prompt you can choose whether you want to mix the entire project with all MIDI objects or just the selected MIDI objects within the current area into one single MIDI object. In this case, the MIDI track and object effects like timestretching, track MIDI transpose, program change, etc. are bounced into the resulting MIDI object.

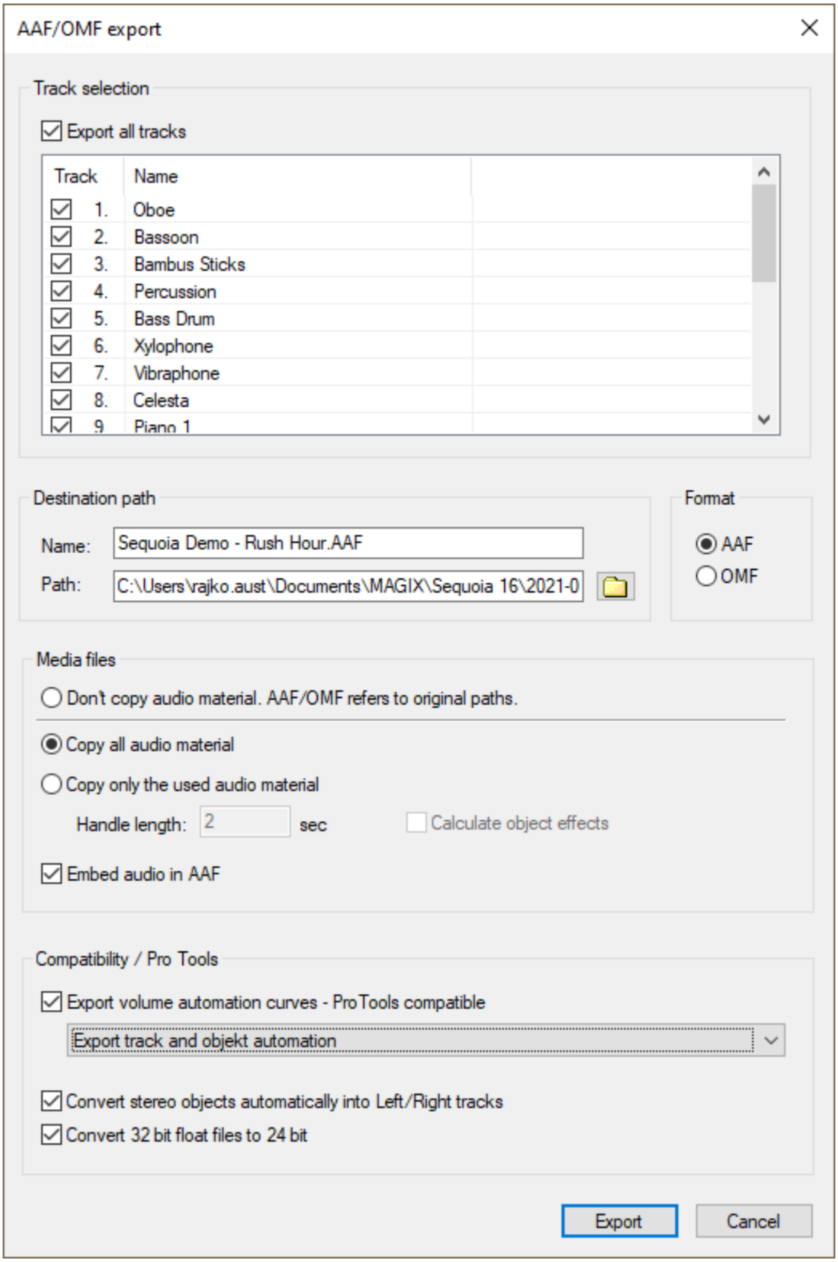


Note: Don't confuse MIDI bouncing with the "MIDI object freeze (Object -> Object freeze) (view page 709)" function, which replaces the MIDI objects with audio objects in the software instrument's audio return signal.

Export project as AAF/OMF

About AAF

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.



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

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

Note: During AAF/OMF export, media files will always be converted to WAV files. You can't export video files. Corresponding objects will not be transferred; the video track therefore remains empty.

Track selection

Select the tracks of the project to be exported. With **Export All Tracks** you select all tracks with one click.

Destination path

Enter here where you would like to export the AAF/OMF file.

Media files

- **Don't copy. AAF/OMF refers to original paths**: If you select this option, no media files will be exported – the AAF/OMF file will only refer to the original media file paths.

- **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. That will be useful when you want to adjust the fades of the objects in the new project.
  - **Calculate object effects:** All realtime effects applied to objects are rendered into the new audio files
- **Embed in AAF/OMF file**: If you select this option when exporting then the whole project will be saved as a single file, which contains both media content and metadata.

> Note: Select the "Embed in AAF/OMF file" or "Copy to destination folder" if you want to have the entire project together with the corresponding Audio/Video files as AAF/OMF files in another studio.

Format

Select whether you want to export the file in AAF format or in OMF format.

Compatibility/ ProTools

Not all static and automated volume values that are exported into AAF are correctly imported into ProTools. ProTools knows only one track volume automation curve and the object volume, called Clip Gain in ProTools. That's why we created some compatibility modes how to handle volume automation when you intend to export your project for ProTools.

Export volume automation curves Pro Tools-compatible:

- **Off:** Object Volume is recognized as ProTools clip gain. Object and track volume automation is ignored by ProTools.
- **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. Object and 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.

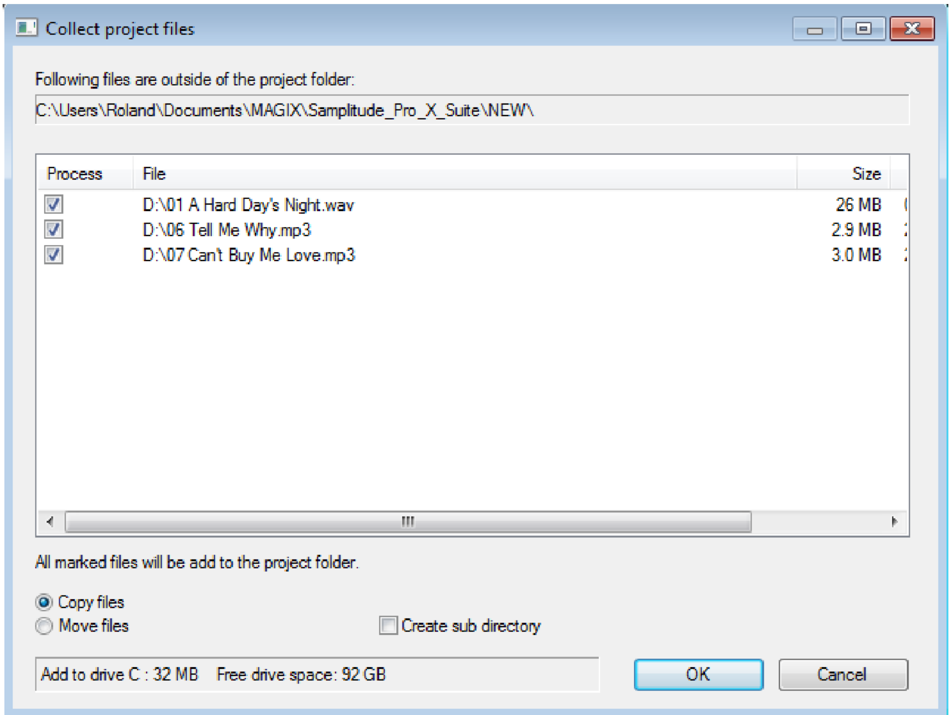
Convert 32bit float files to 24 bit: As AAF uses only 24bit audio files, 32bit float files are converted to 24bit with this option on.

Note: Timestamps saved with ProTools in Frames for improved compatibility.

Clean up

Collect project files

This dialog displays all files located outside of your project folder. All files marked in the "Edit" column can be compiled in the project folder. You can ether copy or move the files outside of the project folder into it or create a new subfolder. Samplitude displays the amount of space required and how much is currently available.



After processing, it's a good idea to save the project!

Delete Virtual Project (VIP)

Use this menu command to easily delete wave projects from the hard disk. The integrated wave files may be individually blocked from deletion by removing check marks under "Delete" next to each file. The remaining files belonging to a virtual project will be deleted.

> **Warning**: Once the dialog has been confirmed, the data will be deleted without a repeat query.

Save Complete VIP in...

The complete virtual project and all associated wave files will be saved in the specified folder. This function is useful for making backups on a different drive, etc.

Keep project subdirectories: If you have already created subfolders in your project directory, e.g. to sort samples, then these will also be copied into the new directory.

Copy unfreeze data: Copies all unfreeze data in the VIP into the new directory.

Copy Revolver track data: Copies all Revolver track data included in the VIP.

Copy ClipStore data (Sequoia only): Copies all of the ClipStore data used in the VIP.

Copy video data: Copies associated video files.

Copy effect used files: Copy the files used by effects into the new directory, e.g. Room Simulator or the vocoder.

Copy only samples used in VIP: Only the sections of wave projects that are actually used by objects in the VIP are copied.

This function is practical for saving disk space; however, the objects in the newly saved project may be only shortened, and not extended. This is because all of the audio data outside the object edges is not copied with it, and therefore, cannot this cannot be used.

A security area in may be defined in samples for this purpose. The **number of samples** will be left as **additional space** in front and at the end of the corresponding object limits to leave a reserve, in case the object may be altered with a fade in/out. A value of 22,050 samples is preset (this corresponds to 500 ms at a rate of a 44.1 kHz).

Burn Project Backup on CD/DVD

Use this menu command to burn an entire project backup onto multiple CDs or DVDs. Samplitude's burner program, MXCDR, will open for this.

Delete Wave Project(s)...

Use this menu command to easily delete wave projects from the hard disk. All files belonging to the HD wave project (the wave file containing the audio files, files with the graphics data, and the HDP file with the project information) will be deleted together.

Note: An HD wave project may only be deleted once it's closed. As long as parts of the project are still active as objects in the VIP, deletion is impossible. Once the dialog has been confirmed, you will be able to delete the data.

Delete Unused Samples

This function lets you edit all wave projects belonging to the current virtual project in such a way that all ranges that aren't used are deleted. It basically deals with the part of the audio data to which the objects in the arranger do not refer and don't get played at any point. The objects in the virtual project are customized automatically so that nothing changes in the arranger view.

This function deletes physical data and does not feature an "Undo" option. You should therefore apply this command carefully.

If multiple virtual projects refer to the same wave files, all of these projects should be open as well. This is the only way to monitor and prevent data loss.

Using the "Remove unused samples" function saves storage space, but corrections to the objects' lengths are limited, since all audio data outside the object borders are removed. For this reason, specify backup reserves with "**Save additional samples for each object**". This number of samples is left in front and behind the object borders in the audio material. The default value is 22,050 samples, which corresponds to 500 ms at a 44,100 kHz sample rate.

The dialog lists all of the wave projects used in the virtual project. You can see the total memory space used by the wave projects in the "Size" column here, and beside this the free space in the "Unused" column. In the "Edit" column, you'll find a check box for each file that may be used to add files to the process. Only those files that contain unused sections of audio data are selected. To keep these, remove the check.

Note: The list may also contain files not actually referenced by the VIP, but referenced by the VIP's undo chain. These files are 100% unused samples and will be deleted completely when checked.

For example, if you have discarded an entire recording session, you can delete the unused files as well. However, if you had opened audio material from other sessions or your private sample library and didn't use them any more after that, you should uncheck these files so that they don't get deleted.

We recommend deleting the "Undo" chain and closing all unused wave projects before using the function.

Tip: If the relevant audio files are required later on for additional production work, the following procedure is recommended for archiving the finished production:

Save your entire project in a new folder ("File -> Save complete VIP in..."). In the save dialog, select the command "**Copy only samples used in VIP**". Now only those audio files/samples that are actually required by your archiving project will be found in the few folder. You can now save the content of this folder to a backup medium (e.g. CD-ROM or DVD).

Delete Freeze Data

Use this function to delete unused data that was created when freezing but is no longer needed after "unfreezing".

Close Project

This menu point closes the active project.

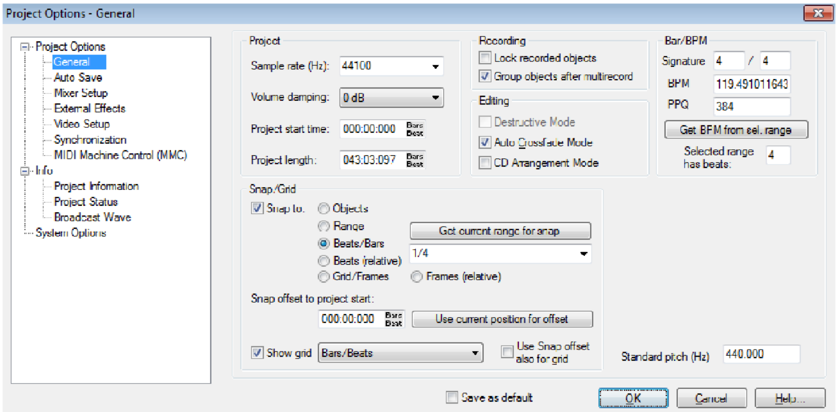
Project Properties

Snap and Grid Setup

In the general project options dialog you can choose the kind of grid and its activation.

Shortcut: I

Virtual projects have an object grid, area grid, beat grid, and frame grid. In the object grid, the objects may only be moved to the start, end or the hotspot (view page 708) ("Object -> Set hotspot/delete hotspot (view page 708)"). The reference point is normally the front edge of the object that is to be moved. As soon as a hotspot has been defined for the object, this is what will be used. If several objects are selected, alignment takes place at the forward edge of the hotspot of the object that was selected last, and is currently located under the mouse cursor. The intervals between the selected projects stay constant.



Grid: Checking the tick box turns snap on globally.

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

Snap to range: Activates the range grid with the option to use the current range as the basis for snapping.

Grid/bars: Activates a grid with bars as the basis. The objects snap to the the next beat grid.

Grid/bars (Relative): Activates a grid with bars as the basis. This means that the relative distances between the objects adhere to the beat grid when you move them about.

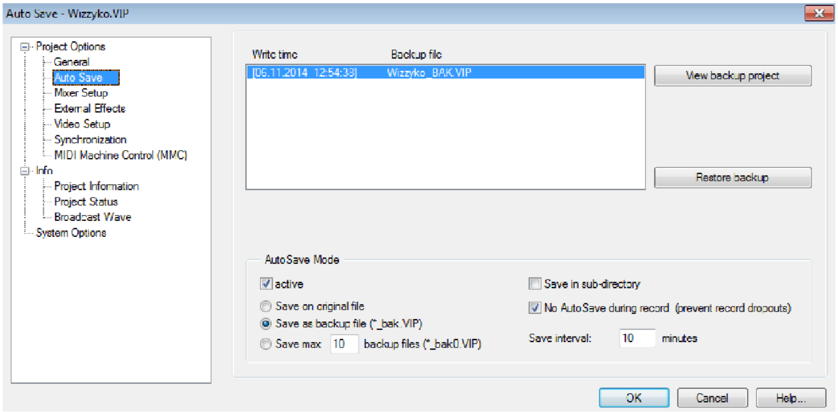
Grid/frames: Activates the frame-based grid.

Snap offset to project start: Here the snap offset can be set relative to the beginning of the project. "Use current position for offset" specifies the current position as the grid's zero position.

Use snap offset also for grid: The snap offset is used as a reference size for the grid.

Save automatically

The **Autosave** dialog is available via the project options.



Here you can adjust the settings for creating backup projects. The button **View backup project** opens the selected backup in the arranger.

The button **Restore backup status** ensures restores the project to the status of the backup. If the current project was changed beforehand and not yet saved, you can save the latest status with the addition of _OLD.vip.

The "Autosave" can be activated by clicking this checkbox. Select between the following options:

Save to original: Overwrites the original file with the current status at specific intervals. If the **Save in sub-folder** option is selected, the previous status, i.e. the last but one status, will also be updated in the **Backup** sub-folder.

Add backup file (*bak.VIP): Adds a backup file and updates this at specific intervals. If the **Save in sub-folder** option is selected, the backup file will be updated in the **Backup** sub-folder each time it is saved.

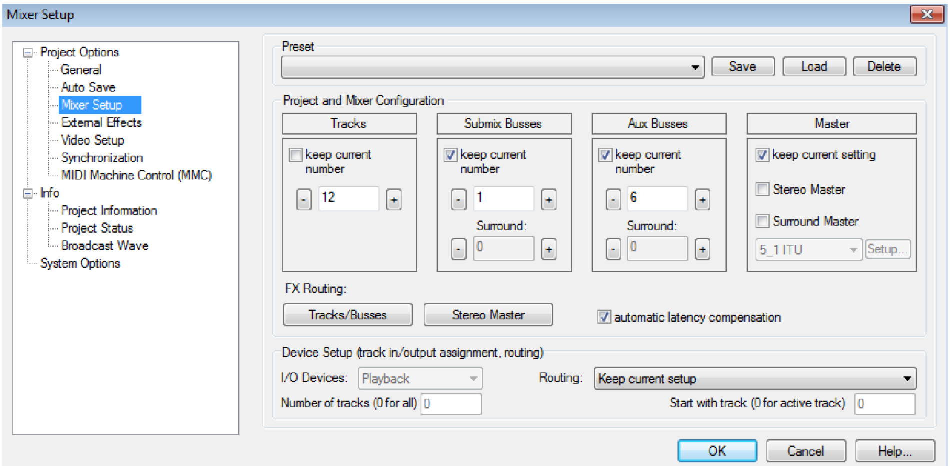
Max. 10 backup files (*bak0.VIP to *bak9.VIP): Specify save intervals for up to 10 backup files and number them. If the **Save in sub-folder** option is selected, the backup files will be updated in the **Backup** sub-folder each time they are saved. After 10 backup steps, the last backup file is given the extension *bak9.VIP: in contrast, the oldest backup file is deleted whereas the others move down in the numbering.

No automatic save during recording (avoid recording errors): Automatic saving during recording is set to inactive to avoid recording errors. Deactivate this option in order to save automatically during recording.

Save automatically after finished recording: When this option is activated, the project is automatically saved directly after the recording has been completed.

Mixer Settings

In this dialog you can easily configure, save and load project and mixer settings.



Shortcut: Ctrl + Shift + M

Project and Mixer Configuration

Here you can set the number of tracks, AUX buses and submix buses. You can also determine the effects and signal routing for the tracks and the master section.

Automatic Latency Compensation: Activate this option for automatic latency compensation of VST plug-ins. This is active by default.

Master Routing

- Keep Current Setting: Each track is assigned to an audio device (or a submix bus). The final mix then takes place in external equipment or on the sound card.
- Stereo Master: Normal mixing in the Samplitude Stereo Master. If the sound card has several outputs, select one in the mixer (in the "out" bar below the volume faders) or in the playback parameters (shortcut: "P").
- Surround Master: If this setting is active, master channels are provided for surround mastering.

Detailed information about the Surround setup is available in "Surround Sound" -> "Surround Setup (view page 298)".

Keep Current Number (Tracks, AUX and Submix Buses)

Check this box to prevent the current settings from being overwritten when loading a mixer preset. The current status of this checkbox is also saved along with a preset. If you quickly want to add a few tracks, (e.g. 4 AUX buses and 4 submix buses), you only have to load the "4 Busses 4 AUXes" preset.

When loading a preset, the respective tracks and buses will be added where the box is not checked. In other words, place a check for settings that shouldn't be included in the preset before saving.

Device Settings

In this section you can conveniently select the input and output device assignment for several tracks simultaneously.

I/O Devices: Here you can define the routing for playback, recording or both.

Number of tracks (0 for all) / **Start with track (0 for active track)**: This option permits a change to I/O routing for a certain number of tracks. If you set the number of tracks to 4 starting at track 8, the selected routing setup is applied to tracks 8-11.

Note: To be able to switch these commands to active, you must first replace the "Keep current setup" option in the neighboring "Routing" field with another option from the selection menu.

Routing

Keep current setup: No routing assignment takes place if a preset is loaded.

Assign all tracks to stereo master: All tracks (incl. AUX and submix buses) are routed to the master. **L/R panning** sets the panorama settings for the tracks to alternate between left and right. Use this option if stereo sources are recorded as mono pairs.

Assign tracks to available stereo/mono devices: All tracks are assigned to the different output devices. In the system options window (shortcut: "Y") you can define which devices are available in Samplitude and which are not. When routing to mono devices, the tracks are also placed right/left in panorama alternatively.

Assign tracks to surround channels: The corresponding tracks are created here after the Surround setup has been selected .

Use setup from preset: The settings saved in the preset are used.

Video Setup

The integrated video engine allows you to add video objects to the first track. Video objects can also exist in other tracks, but they will not be played back, and will appear shaded. Various video files like AVI, DV, MOV, MPEG, MXV can also be loaded. WMV or image files such as BMP or JPEG can be loaded into the arranger window using drag & drop or through "File > Import... > Load Video file" (view page 572).

Make sure that the video files included in a project are identical in terms of dimension, aspect ratio, and frame rate. If possible, use only videos that work exclusively with keyframes and without interframes (DiVX or MPEG are not very suitable). If necessary, you can convert files into other formats and resolutions using the video editing program "Movie Edit Pro". We recommend the DV and MXV formats.

The video soundtrack can be loaded into the project. This process converts MPEG-2 audio into WAV format.

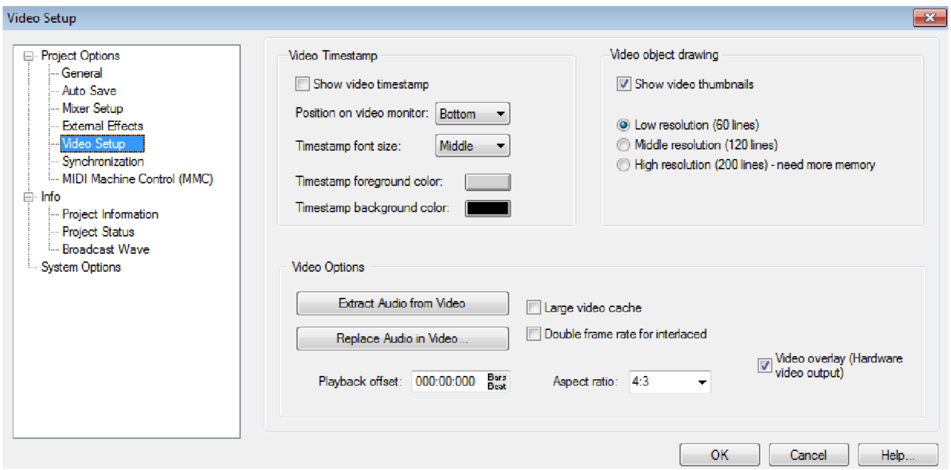
You can move or cut video objects just like all other objects, but no fades will be used. It is possible to edit a video using Source/Destination cut commands.

When working with many single bitmap files, keep in mind that these will be loaded completely into RAM and not successively read from the hard disk.

You can **right-click the video object** to

- open "Video Settings" dialog window
- show/hide the video window
- replace the video file
- move/shift the video object by entering values

Video Setup - Dialog Options



Show video time stamp: This option opens the time stamp display in the video monitor. You can display it in three sizes either at the top, in the center or at the bottom of the project window.

Video object drawing: The option "**Show video thumbnails**" displays thumbnails in the video track in different resolutions. You can improve system performance by unchecking the box to shut off the video object display completely. The video file will still be played in the video monitor. VIP draw mode 2 ("Tab" key) can be used to display only the object start and end frames.

Extract audio from video: This button extracts and converts audio files from an AVI file into a wave file for insertion into the current project.

Replace audio in video: This option can be opened using the "Export video sound" function to replace the existing audio track in an AVI video.

Playback offset: Set a positive or negative playback offset in this field.

Large video cache: This option allows you to allocate more cache for the video clipboard.

Double frame rate for interlaced: This enables you to double the image repeat frequency when interlacing so that videos can be viewed without any flickering.

Aspect ratio: Select an aspect ratio of either 4:3 or 16:9.

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).

Synchronization Setup...

Shortcut: Shift + G

Read "Synchronization (view page 450)" for more information about synchronization settings.

Synchronization Active

This command activates synchronization.

Shortcut: G

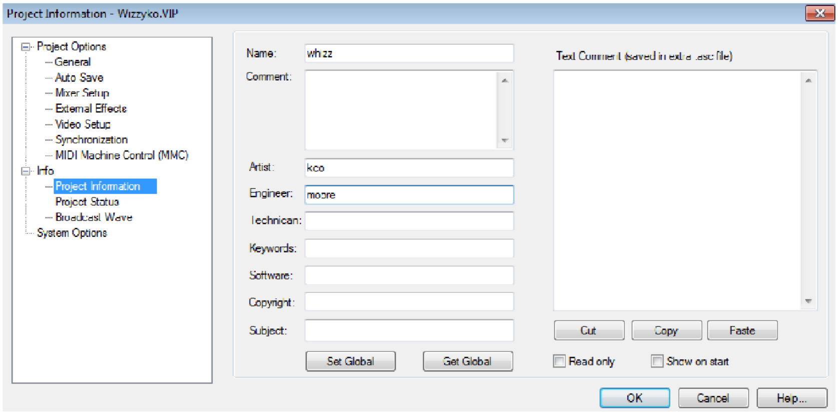
MMC Setup...

Read "Synchronization >MIDI Machine Control (view page 456)" for more information about MMC.

Project Information...

Enter information about the project here.

There is also a text field for comments. These may be set to show each time the project is opened.



Project Status...

This indicates all project status information like the name, path, number of ranges, markers and objects, wave files used in the current project, or the date of creation. You can also create a text file which includes all of this information.

Broadcast Wave Manager

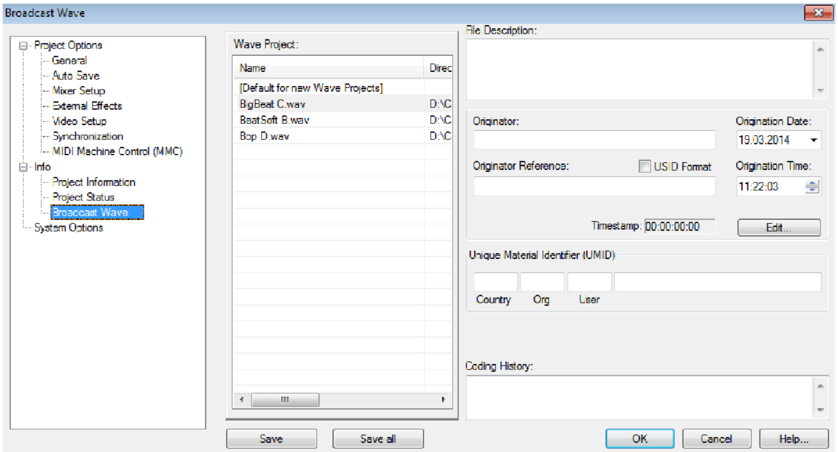
The Broadcast Wave Extension makes it possible to save information about the audio files as metadata in the extension of the Broadcast Wave Format (BWF) called the "Chunk". This metadata can basically be applied in a proprietary manner, but observing the respective EBU and SMPTE guidelines is recommended. The file ending of the Broadcast Wave format remains ".wav".

Wave Project

Here you'll see a list of all the WAV files in the project. To add or read specific information, simply click on the corresponding file. Its current BWF information will be displayed in a dialog on the right.

For newly recorded audio material you can define the metadata that should be written to the Broadcast Wave Extension.

This metadata can late be read or extracted and used for other purposes such as the management of audio files in a database.



Broadcast Wave Manager - List of individual fields

All of the settings values are saved according to the project and used for new audio files.

File description: You can use this field any way you like. You can enter a text of up to 256 ASCI characters.

Originator: This field contains details on the origin of the file, e.g. the description of the producer. Maximum of 32 characters.

Originator reference: This field is determined by the originator. This can be an internal reference number, for example. In the area of EBU, the EBU Recommendations R 99-1999 define how this field should be completed. To assign the entry of this property, activate the "USID Format" option. Afterwards you can format the entry according to the EBU recommendations. Maximum of 32 characters.

Date: The date of the file's creation is displayed and can be edited, e. g. if audio material was saved for the first time as a file, even though the recording is somewhat older and the date needs to be valid as a reference. If the entry is processed in the BW Manager, a value will result that is independent of the file properties.

Time: The time at which the file was initially created is shown here. As with the date, this is automatically created from the file properties but can be edited retroactively.

Only Sequoia: ISRC: Enter the ISRC of the file here. This is a 12-digit ID number that provides specific information like the originating country of the label, the label's company number, the year, and a sequential title number. ISRCs are used by the music industry for the identification and accounting of music titles.

Timestamp: The timestamp saved in the BWF extension is displayed here. This is the timecode for recording the file, which is identical with the timecode of the first recording if the recording was synchronous. For other applications, this timestamp can provide information on the time on the day of recording. Click on the "Edit" button to adjust the timestamp or to apply it from the object position. Here all audio files are assigned the timestamp from the object positions or the timestamp assignment is undone.

Unique Material Identifier (UMID): The specifications for UMID are regulated by the SMPTE (Society of Motion Picture and Television Engineers). The corresponding documentation has the code number SMPTE 300M-2000. We recommend observing these guidelines and agreements regarding the use of the UMID before using this feature, especially the sections that specifically apply to how you want to use it. Use of UMID is not absolutely necessary for a valid BWF.

Only Sequoia: Loudness: Here you can calculate the loudness metadata as BWF files or update it automatically.

Coding history: In addition to information about the file format (A: encoding, e. g. PCM, F: sample rate, W: bit width, these values are only used with non-transparently encoded material like MPEG or MP3, M: number of channels), each entry in this field contains a T value. This is a comma-free text string for entering the serial numbers of analog tape recorders, codecs, dither types, AD converters, or special signal editing applied to the file (such as de-noising).

An entry is made when a file is recorded in Samplitude. If this file is processed again (e. g. by bit width reduction or MPEG encoding), an additional entry is added.

Use of the coding history is regulated in the EBU Recommendation R98-1999.

CD Arrange Mode

If this is activated, Samplitude arranges recently added objects to insert a Red Book Standard-compatible pause between the objects.

> Detailed information about "CD arrange" mode is provided in "CD arrange mode (view page 1008)".

Offline Audio Editing (Destructive Wave Editing Mode)

This menu item activates or deactivates offline audio editing in the wave project window. If this feature is deactivated, Samplitude reverts to "Virtual Wave Editing" mode.

Detailed information about destructive editing is available in the chapter "Working in the Project Window > Audio Editing in Samplitude (view page 66)".

Program Preferences

System options

Specify all of the important settings for Samplitude in this dialog.

These include:

- **System options**: Audio setup, audio devices, MIDI, hardware controllers, metronome, record, play, performance, advanced buffer settings
- **Program preferences (program)**: General, undo, object locking
- **Keyboard, menu & mouse**: Keyboard/menu, MIDI editor, special keys, mouse, mouse wheel.
- **Design**: Skins, display options, colors
- **Effects**: Dithering, resampling/bouncing, VST/ReWire, destructive effect calculation
- **Option administration**: Reset program settings to default values
- **Project options**: Switches the tree structure of the project options

Explanations and references regarding system settings are located directly in the "System options" dialog.

Audio Setup

Driver System

Please also read "System settings -> Audio setup (view page 70)" for more information.

Buffer Settings

Please also read "System settings -> Audio setup (view page 70)" for more information.

Device/Driver Communication

Detailed information about this topic can be found in the chapter "System Options" > "Audio Setup" (view page 70).

Monitoring settings

There are different options for monitoring available here. These settings are used as presets for new projects.

For more information about monitoring, please see "System settings -> Monitoring settings (view page 72)" for more details.

Audio Devices

Detailed information about this topic can be found in the chapter "System Options" > "Audio (view page 81) Devices".

MIDI

For more details about MIDI settings, see "System settings -> MIDI settings (view page 81)".

Hardware Controller

Detailed information about this is available in "Hardware controller".

Metronome options

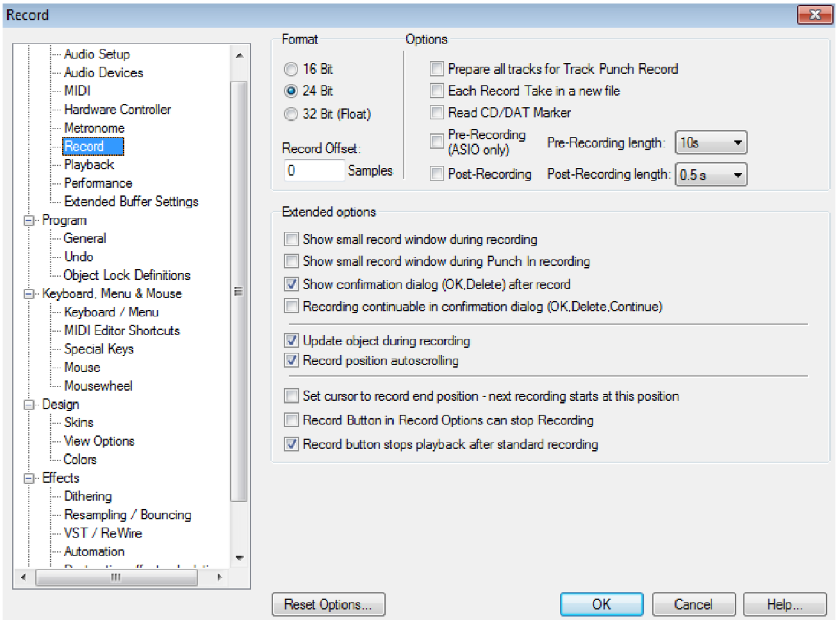
Detailed information about this is available in "File -> Program Preferences -> Metronome Options (view page 44)".

Recording

This dialog enables the selection of various dialog configurations and playback marker behaviors when recording.

Format: Choose from 16-bit, 24-bit or 32-bit (float).

Enter the **Recording offset in samples** to ensure consistent movement of your audio recordings with regard to the existing audio material in the arrangement.



The following **options** are available:

- **Prepare all tracks for track punch record**

If this option is activated, all set record devices will open when the playlist is opened, so that you can start a recording on all tracks using the track record button.

- **Each record take in a new file**

This option results in each take being saved to a new file. Loop recordings will be processed as one take and saved to one file.

- **Read CD/DAT Marker**

DAT devices and several professional CD players output digital marker information via an SPDIF output e. g. CD track markers or DAT markers). With this recording option all marker information is read from the soundcard's SPDIF input, provided the selected audio device supports it.

Pre-Recording (ASIO only): If recording is activated from the stop state or during playback, this function inserts audio material that you have added at the beginning of the recording to the beginning of the current recording. You can set pre-recording lengths of 2, 5, 10, 30, 60 and 120.

If you drag out a recorded object to the left, you'll see the set pre-recording time before the start of the actual recording.

Post-Recording can also be activated in the recording options in order to record up to 2 seconds of audio material in the background after the actual recording is ended. If you drag out the object end of the recording to the right, you'll see the display of the part that was recorded after it was stopped. The preset for post-recording equals 0.5 seconds.

Extended options

Show small record window during recording: Displays a small non-modal recording window with the most important recording controls while recording.

Show small record window during punch recording: Displays a small non-modal recording window with the most important recording controls during punch recording.

Show confirmation dialog (OK, Delete) after record: If this option is checked, a dialog will appear after the recording process to choose whether to keep or discard the recording.

Note: If you deactivate this option, recordings will always be saved straight after they have been completed. On the one hand, the best security against data loss as a result of inadvertent deletion is guaranteed, and on the other hand, obviously unnecessary takes are saved to the hard drive and take up space there until they are manually deleted.

Recording continuable in confirmation dialog (OK, Delete, Continue): With this option, when you click on the record button during a running take, a dialog will appear where you can choose to delete, stop or continue the recording.

Update object during recording: This option increases the recorded object constantly along with continuous recording.

Record position autoscrolling: If this option is activated, the window will scroll along with the recording cursor.

Set cursor at the end of recording - next recording starts at this position: If this option is activated, the cursor will be set at the end of the recording and playback or recording will start at this position.

Recording button in recording options can stop recording: Here you can specify that the **"Record"** button in the **Recording Options** (shortcut: **Shift + R**) is able to start or stop the recording process.

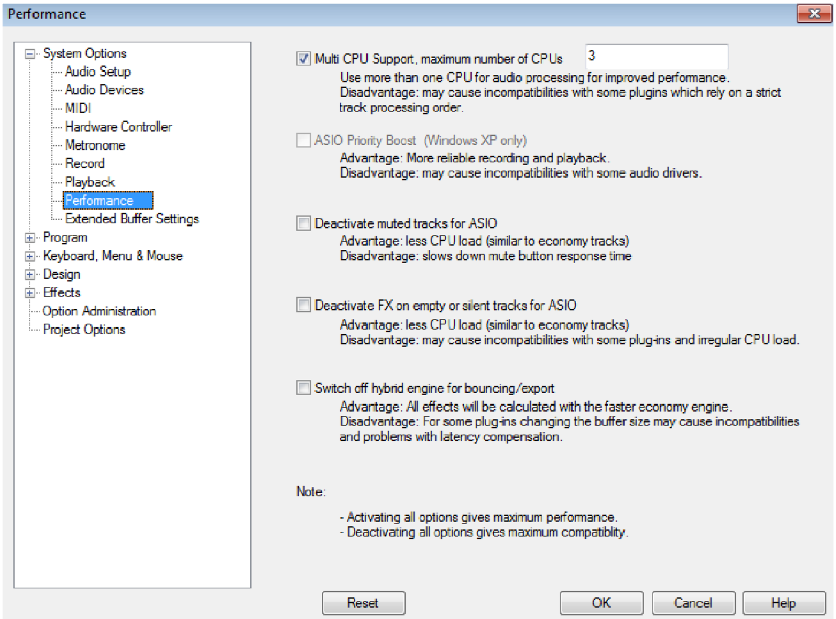
Record button stops playback after standard recording: If this option is activated, the running recording and playback can be ended by clicking the "**Record**" button in the transport console.

Playback Options

More information is available in "Playback -> Playback parameters (view page 727)".

Shortcut: P

Performance



The options featured under "Performance" are provided to set up your system's performance optimally. Among other things, "**ASIO silence economy**" is available for switching off empty or silent tracks with ASIO.

Multi-CPU support: Samplitude supports use of multi-CPU/multi-core/HT workstations that distribute the load of various tasks across several processors. If you select the driver system "MME" or "WDM", the audio thread editing will run mainly on the first CPU. Disk I/O operations, graphics, and video integration are processed by additional CPUs.

"ASIO" activates this driver system and dual CPU support; the mixer tracks and their effects are distributed between the initial CPUs. Objects (including their effects), integrated video material, and the graphics engine are distributed between additional available CPUs. The program is currently optimized for work with up to eight cores; working with 3 cores is preset.

If you own a system with multiple CPUs, you will be able to get higher performance from your system for processing audio, provided that you activate this option. However, using multi-CPU support can lead to incompatibilities with plug-ins that are specified for a set track processing series.

ASIO priority boost: Among all threads of the application, special priority is granted to the ASIO thread. This option is a preset and does not need to be modified. This increases the reliability of your audio recordings as well as that of the playback.

Note: We recommend deactivating this option for LYNX and TASCAM devices.

Deactivate muted tracks for ASIO: Use this option to reduce the CPU strain on your system. When muted, however, there may be some delays.

Deactivate FX on empty or silent tracks for ASIO: Use this option to reduce the CPU strain on your system even more. This function can, however, cause fluctuating CPU efficiency.

Switch off Hybrid Engine for bouncing/export: This option calculates all effects during the bouncing/exporting process using the faster "Economy Engine". This may cause irregularities when some plug-ins attempt to compensate for latency.

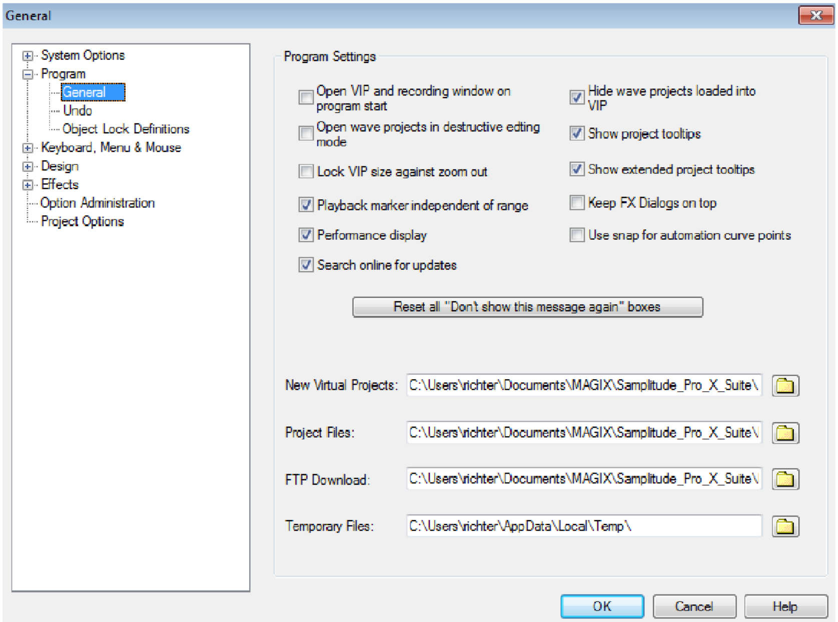
Note: If every option is activated, maximum system performance will be achieved. If each option is deactivated, maximum compatibility will be achieved.

Advanced buffer settings

Detailed information about this is available in "File -> Program Preferences -> Advanced Buffer Settings (view page 622)".

Program

General options



Open VIP and record window on program start: If this option is active, Samplitude will automatically open an empty virtual project (VIP) and the "Recording parameters" dialog when the program is started.

Open wave projects in destructive editing mode: If this option is activated, audio files are opened in offline editing mode.

Lock VIP size against zoom out: If you zoom the virtual project out beyond the section size, this option will automatically stop it from being extended.

Playback marker independent of range: This option allows the playback marker to be set independently in "Loop" mode and playback to be started outside of the selected loop range.

Performance display: This option activates the DSP performance display in the lower left corner. Your PC is overloaded if the displayed rate is 100% or higher. If this is the case, try one of the following steps:

- Reduce the number of realtime effects in the mixer or in the object editor.
- Reduce the number of tracks by muting tracks that are not needed.
- Increase the VIP buffer size (keyboard shortcut: Y) to 16000 or 32000 samples.

Please note the comments displayed in the status bar if an overload occurs during realtime previewing of complex effects such as the DeNoiser or FFT analysis filter.

Search for online updates: When this option is activated, Samplitude will automatically search for updates each time it is started.

Create fades for new objects with Auto Crossfade mode active: When loading audio files into the VIP, the objects will be created without automatic fades even if Auto Crossfade mode is active, as it is assumed that such samples have already been cut properly. If you still want fades in this case, enable this option.

Show project tooltips: If this option is active, additional information is displayed when the mouse pointer hovers over an interactive element.

Show extended project tooltips: If this option is activated, the extended tooltips are displayed if the mouse pointer is held over an interactive element.

Keep FX dialogs on top: This option keeps all effect dialog windows "on top" of all other windows.

Use snap for automation curve points: Select this option to set automation curve points while moving set values on the grid.

Reset all "Don't show this message again" boxes: Click this button to reactivate all tips dialogs that have been deactivated.

Path settings

New Virtual Projects: All new virtual projects as well as recorded and imported wave projects will be saved through this path.

Project Files: All other wave files which are to be saved on the hard drive and cannot be allocated to a specific project are saved through the specified path.

FTP Download: All files downloaded via the integrated FTP client will be saved through the specified path.

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

Undo

Detailed information about this is available in "File > Program Preferences > Undo Definitions (view page 624)".

Locking objects

Detailed information about this is available in "File -> Program preferences -> Lock Definitions (view page 625)".

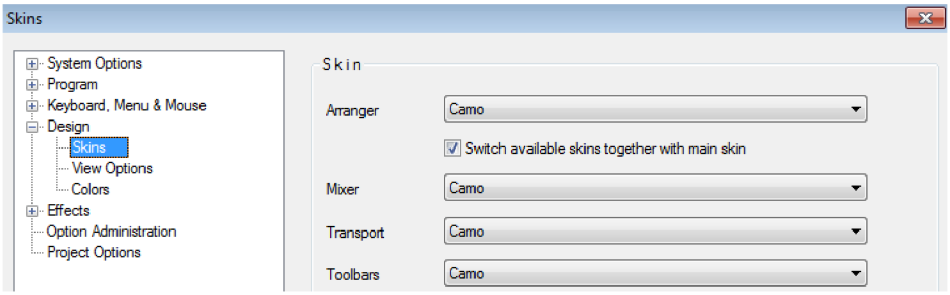
Keyboard, Menu & Mouse

Detailed information can be found via "Menu File -> Program Preferences -> Edit Keyboard Shortcuts and Menu (view page 626)".

Designs

Skins

In this dialog, select different skins for the project, the mixer, and the transport console. Use the various skins to adjust the color and graphical display of the Samplitude interface.



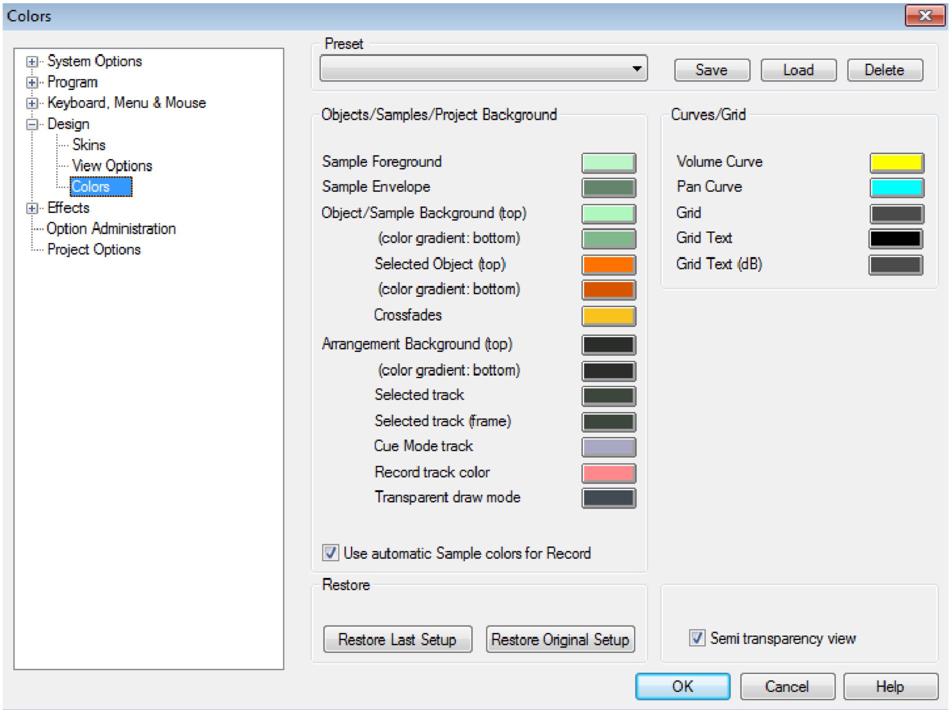
Various skins may be selected by clicking on the icon to the top left in the title bar of the Mixer (view page 204) or right-clicking in the Toolbar of the Arranger.

Project Display

Detailed information on display options can be found in "File > Program Properties > Display Options (view page 631)".

Colors

All colors used in the program can be set here. You can also load and save these colors as presets.



Use automatic sample colors for record: A separate random color is chosen for each recorded object per track.

Redo

Restore last state: The last status of the color settings prior to opening the dialog was opened is restored.

Reset...: Reset the color settings to those of the previous or original color state.

Previous state: The previous color settings will be restored.

Original state: Resets all colors to their default setting.

Reset object colors: This option resets all object colors. This may also be necessary when importing a project from a different PC with different color settings, or if the objects have colors that deviate from the standard object colors as a result of automatic assignment during recording.

Reset track colors: Resets the track colors.

Reset object colors to track colors: Sets the object foreground colors of the waveforms to the track colors.

Object background to track colors: Sets the object background colors to the track colors.

Semi transparency view: Some effect dialogs and editors feature transparent displays. These may be deactivated here to increase performance.

Effects

Dithering

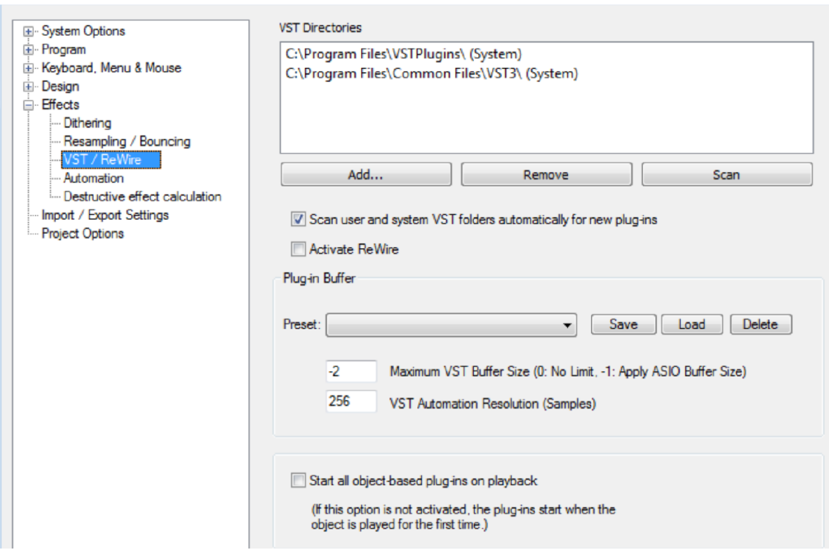
You can find out more about this by reading "Dithering options..." (view page 638).

Resampling / Bouncing

Detailed information about this can be found in the section „Resampling / Freeze Options“ (view page 640).

VST//ReWire

Keyboard shortcut: "Y" > "Effects" > "VST/Rewire"



VST Plug-in folder list: Set the paths for your VST plug-in effects and VST instruments here. You can add or remove search paths for VST plug-ins with "Add" or "Remove". With "Scan" Samplitude will scan the selected folder for VSTs. Not only are all the plug-ins imported, but they are also checked for usability within Samplitude. More information regarding this dialog and general information on installing VST plug-ins can be found under Installation of VST Plug-Ins (view page 411).

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

Activate ReWire: If this option is activated, rewire-compatible client applications can be integrated into Samplitude as synthesizers.

ReWire is generally used for sample-exact, real-time transfer of audio channels between two programs. Both programs can be connected through the same sound card. The transport functions of the applications (such as playback and forward/reverse) are connected with ReWire. If ASIO drivers are being used, sounds from ReWire-compatible applications can be assigned to different sound card outputs.

To integrate ReWire applications into Samplitude, find the ReWire-compatible instrument that you want to connect with Samplitude in the "MIDI out" slot of the track editor by going to "New Instrument > ReWire"   The instrument outputs of the ReWire application can be opened in the "Audio in" slot of every track.

The tempo is always configured according to the settings specified in Samplitude, i.e. Samplitude is the master. The tracks/channels connected with ReWire-enabled software can be edited with the equalizer, effects, and other plug-ins and then routed to the available buses.

Note: Make sure that the set sample rates of the two applications connected with ReWire match one another so that playback at the correct pitch can be guaranteed.

Additional information about ReWire is available in "ReWire client applications" (view page 427).

Plug-in Buffer

The default presets usually cover most applications when working with plug-ins. Try out various presets if you are experiencing problems with plug-ins or DSP cards.

You can enter the maximum buffer size that will be applied to VST plug-ins. The following special values apply in this case:

0: The program sets the buffer size.

-1: The ASIO buffer size is used

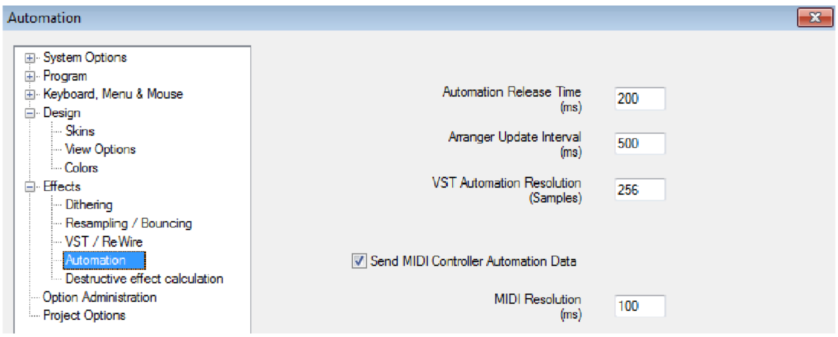
The VST buffer size is equal to the standard VIP buffer size. To use the ASIO buffer size in the Hybrid Engine, enter the value "-2" or select the "Forced ASIO buffers (Hybrid)" preset. The "Forced VIP buffers (use UAD & Powercore in economy engine)" setting diminishes latency with UAD/Powercore plug-ins.

VST automation resolution

The Hybrid Engine uses the ASIO buffer size. To achieve shorter automation times in the "Economy Engine" (which works with VIP buffers), enter a corresponding lower value here.

You can also activate the "Start all object-related plug-ins at play start" option. Some plug-ins cause delays when switched on; in this case, these plug-ins should be activated before starting playback, otherwise errors may occur when the corresponding objects are played back.

Automation



Automation Release Time (ms): This sets the fader return time in ms for the write automation modes.

Time for entries by users: As soon as a controller curve is updated (i. e. new controller data for the curve is received). If no further new data is received, Samplitude transfers the original data again after this time span has elapsed.

Arranger Update Interval (ms): When mapping controllers you can set the frequency of the display update in milliseconds in this field.

VST Automation Resolution (samples): Time constant (in milliseconds) for sending current controller values: When starting playback, the controller data is transferred after this time has lapsed if its value changes (within the frame of 7 bit MIDI resolution). During playback, the VST controller values are exported at this rate and sent to the corresponding VST plug-in. During playback the VST controller values are retrieved within this time gap and are sent to the respective VST plug-in.

Note: VST2 does not support sample-exact automation, only parameter changes at buffer limits. To increase the automation resolution enter a low sample value here. Lower values can lead to a higher load on the CPU for automated effects.

Send MIDI Controller Automation Data: Here you can globally activate or deactivate the sending of MIDI controller data. Program-internal automation is not affected by this.

MIDI Resolution (ms): Time constant (in milliseconds) for sending current MIDI controller data. The controller data is sent during playback after this time period if the value changes (within the frame of the 7-bit MIDI resolution). During playback the MIDI controller data is retrieved and sent to the respective MIDI device.

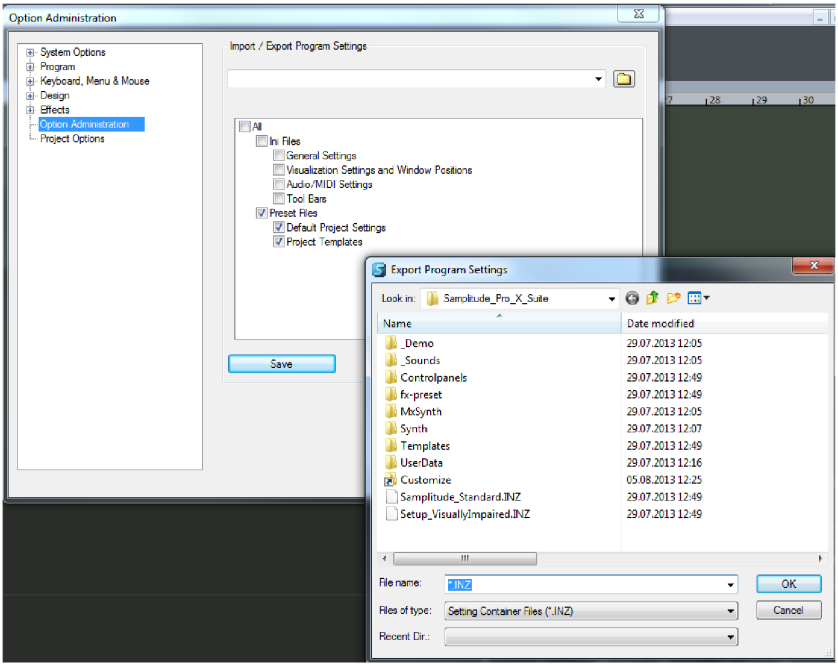
Destructive Effect Calculation...

Choose in the advanced destructive effects calculation options (view page 757) if you want to add the effect to the original file when editing destructively, write it into the effect file, or create a new effect file for each calculation.

Options managerment

The Samplitude folder includes so-called "Ini" files and preset files, which contain program settings.

The following dialog enables the listed settings to be saved and loaded conveniently. This makes it possible to save the complete program settings onto a mobile storage device for use on other Samplitude workstations.



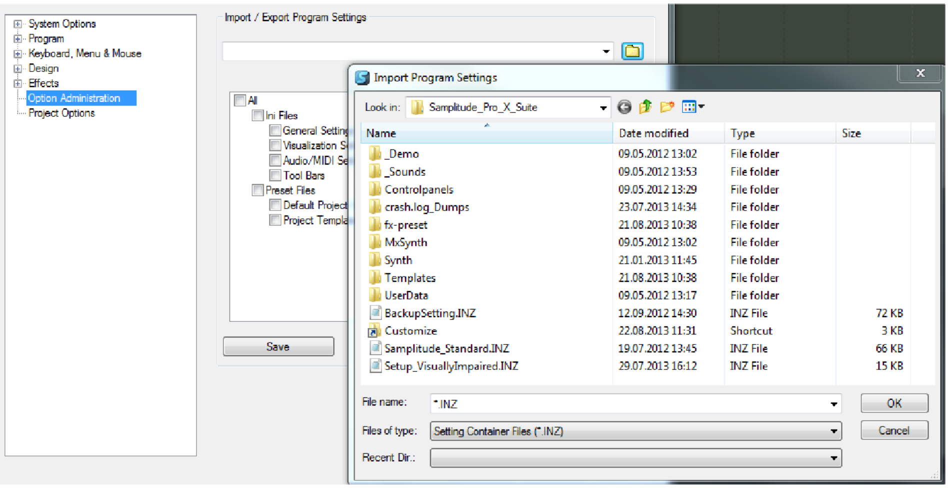
The program settings are saved as "**Settings Container Files (*.INZ)**".

Save: Press the "Save" button to save the current project and program settings in a "Settings Container". All of the program settings will be saved, including those that are not checked. The state of the individual check marks during saving only represents a pre-selection for loading. The saved file will now appear as a preset at the top of the list.

Samplitude settings are provided for the following dialogs:

- General settings (view page 611)
- Settings displays and window positions (view page 631)
- Audio/MIDI Options
- Toolbars
- User administration (affects all users and user settings) (Sequoia only)
- Standard project settings (view page 84)
- Standard Crossfade-templates (Sequoia only)
- Project templates (view page 574)

Load: Select a program setting for Samplitude from the list above or import the settings via the folder button.



Specify via the tree structure which "Ini" files or "Preset Files" for the selected container file should be used. Now press "**Load and restart**" to activate the settings. Samplitude will restart and a backup of the last settings will be created.

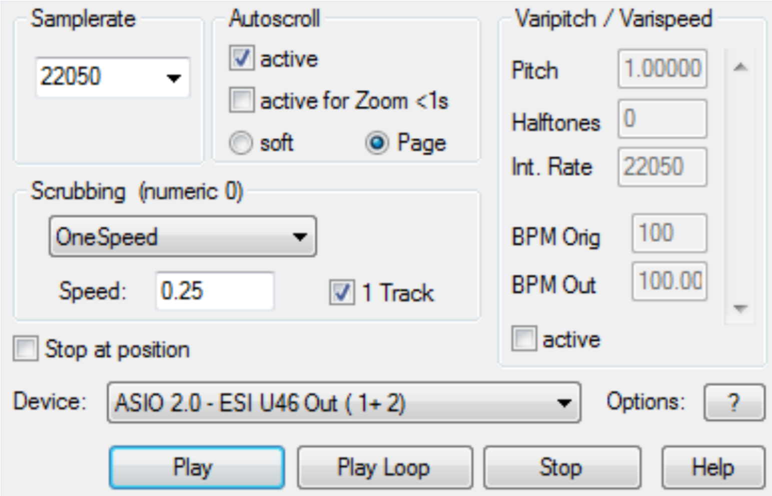
Hint: Load your settings containers comfortably from the list box in the Start Wizard (view page 1042)!

The program **subfolder "Customize"** also features a series of "**INI patches**" that do not contain complete files, but rather only activate or deactivate individual settings. These may also be loaded just like a "Settings Container" via the folder button. After selection is confirmed, a separate dialog window featuring explanations about the respective "INI patch" will appear.

Playback

More information about playback settings is available in "Playback -> Playback parameters (view page 727)".

Varispeed / Scrub Settings



In this part of the dialog you can make changes to the playback tempo for virtual projects. Right-click the scrub control button in the transport controls.

Sample rate: The preset project sample rate is displayed here. If you want to change this value, a dialog window appears where you can optionally adjust all audio objects to the new sample rate. The customizing is conducted, if necessary, through moving and resampling. MIDI objects can also be adapted to the new sample rate. These are also moved if necessary without changing the musical position.

Autoscroll / Scrubbing: See "Playback Options (view page 727)".

With **Stop at current position**, the playback marker stays at the current position when the "Stop" button is pressed and doesn't jump back to the original position.

With "**Device**" you can specify the driver of the sound card that you want to use for playback.

Varipitch / Varispeed Mode

Vertical slider: This controller defines the playback from -200% to +200%, in which case negative values cause the project to play in reverse.

Pitch: Double-clicking on this field enables the speed factor to be input directly, e.g. 0.5 for half speed, 2.0 for double speed.

Halftones: This field specifies pitch changes according to semitones; a value of -12 causes the project to play an octave lower at half speed, and a value of +12 plays back the project one octave higher at double speed.

Internal rate: Here you can specify the sample rate for the Varispeed calculation. At a sample rate of 44100 kHz, a value of 22050 plays the project an octave lower at half the speed.

BPM orig: Displays the original speed in BPM.

BPM out: Enter the target speed in BPM. Samplitude calculates the Varispeed pitch factor from the quotient from the target to the original tempo.

Active: Checking this box activates Varipitch/Varispeed.

MIDI Options...

More details about MIDI settings are provided in "System settings -> MIDI settings (view page 81)".

Metronome Options...

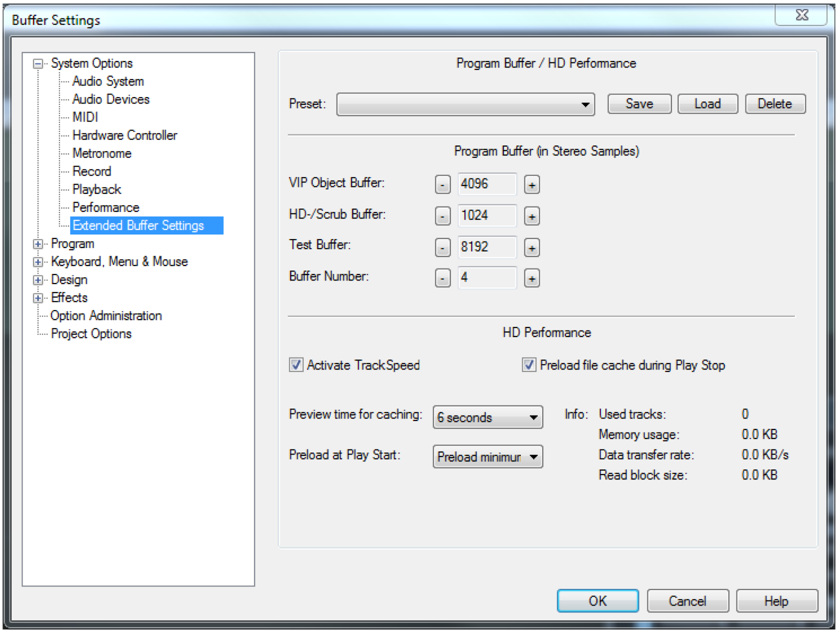
Detailed information about metronome settings is provided in the chapter "Samplitude quick start -> Workshop: recording -> Metronome settings (view page 44)".

Advanced Buffer Settings

In this dialog you can optimize buffer settings for your virtual project, for hard disk caching and for plug-in manipulation. If you are not having any problems with audio dropouts or pops and clicks, you do not need to make any adjustments here.

Program Buffer/ Hard Disk Performance

For special applications you can set and save your own buffer settings. In addition to this, special presets are available, e. g. for scrubbing and ASIO applications.



Program Buffer (in stereo samples)

VIP Object Buffer: Because error-free playback is usually more important than fast reaction times, this value should be increased when playing lots of tracks. This is the only setting relevant for playback and editing in the VIP.

HD/Scrub Buffer: This buffer is put into use during direct playback of HD Wave Projects. Test smaller values here for even faster response times.

Test Buffer: This buffer is only used for realtime previewing of effects from the effects menu.

Buffer Number: Specify how many of the buffers described above should be used here. More buffers increase reliability, but also increase memory requirements. This results is longer response times. You can check the current buffer usage levels during playback in the status bar at the bottom right. We recommend using between 4 and 6 buffers.

Hard Disk Performance/TrackSpeed

With the help of the TrackSpeed technology you can increase the number of hard disk tracks simultaneously playable in Samplitude. TrackSpeed uses your PC's RAM to intelligently and effectively preload the necessary audio data.

By using TrackSpeed, the internal audio engine of Samplitude can even work using multiple tracks with small audio buffers (e. g. 8000 samples or less), without hurting performance. This makes short reaction times between user interactions possible, e. g. with the mixer and various object handles.

There are settings available which can be used to customize TrackSpeed to your individual requirements:

Activate TrackSpeed: Set a check mark here to activate TrackSpeed.

Preload file cache in "Stop" state

If this option is activated, the file cache will be preloaded each time the playback marker is repositioned. You can recognize this by the "preloaded cache" notification that appears in the bottom right status bar. The next play start happens automatically because the necessary audio files are already present in the RAM.

Preview time for caching: Enter how many seconds of the VIP can be preloaded in this field.

Preload at play start: This value determines how much of the cache will be loaded at play start (in percent). Larger values delay the start process but increase the playback stability when using a large number of tracks.

The dialog also displays the following project information:

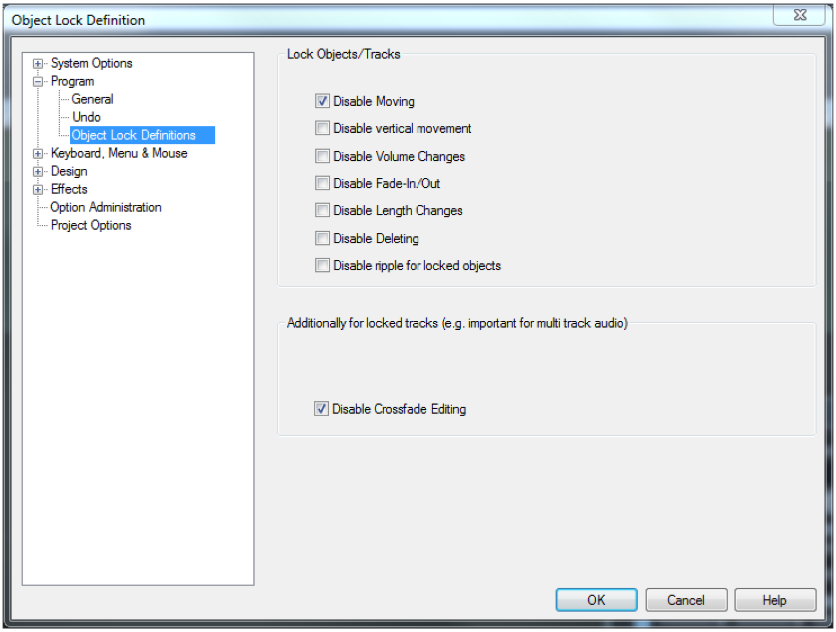
- Used tracks:
- Memory usage:
- Data transfer rate:
- Read block size:

Undo Definitions...

Use this editor to switch the undo function on and off for virtual projects, wave projects, and plug-in settings. The amount of undo steps may also be entered. A value of 20 means that the last 20 changes to a project may be undone.

If you check the corresponding box, temporary undo files be created as HD wave projects for RAM wave projects.

Locking options



Select which functions should be disabled by locking objects or tracks here. You can select the following options:

Locking objects

Disable moving: Prevents objects from being moved horizontally. Vertical movement between the tracks is still possible.

Disable vertical movement: Prevents objects from being moved vertically.

Disable volume changes: This deactivates the object's volume handles.

Disable fade-in/out: Deactivates the fade handles of the objects.

Disable length changes: Deactivates the length handles of the objects.

Disable deleting: This option prevents the deletion of locked objects.

Disable ripple for locked objects: Locked objects are not affected by functions like "Edit -> More -> Delete with time/ripple" or "Edit -> More -> Cut with time/ripple".

Locking objects/tracks

Note: Tracks are locked in the track box by activating the lock symbol.

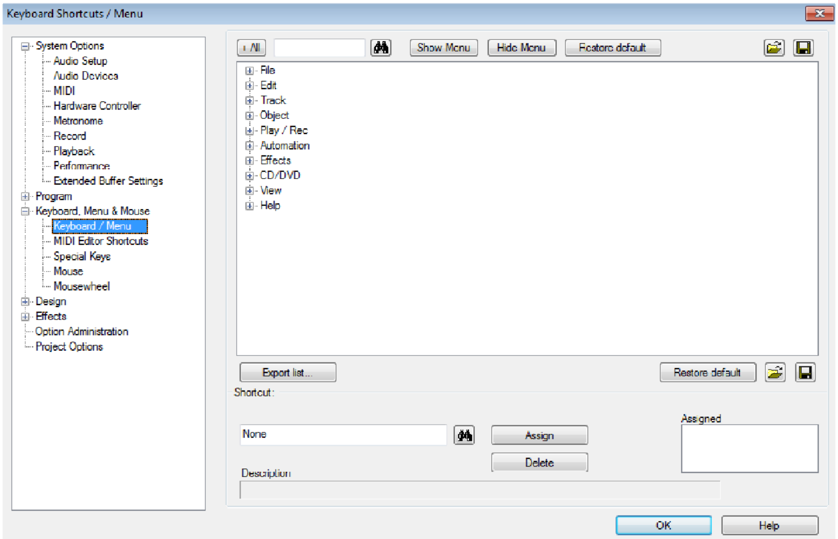
Disable crossfade editing: This deactivates the crossfade function for set tracks. The crossfade can no longer be changed.

Crossfade Editor Preferences

This function has been revised in Samplitude. For the latest information, see the PDF document **Samplitude Pro X7 New Features** in the program folder.

Please see "Crossfade editor" for more information about these options.

Editing Keyboard Shortcuts and Menus



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

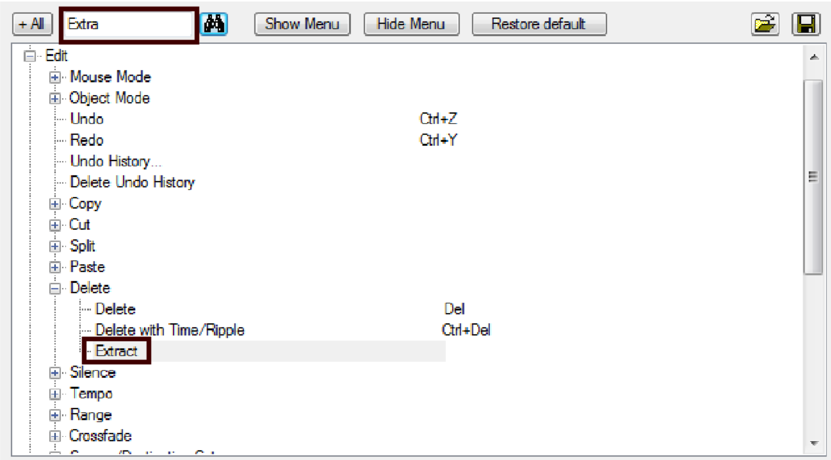
You can also hide menus that you don't really use.

When you close Samplitude the shortcut and menu settings will be saved automatically, so you can continue using them when you next open the program.

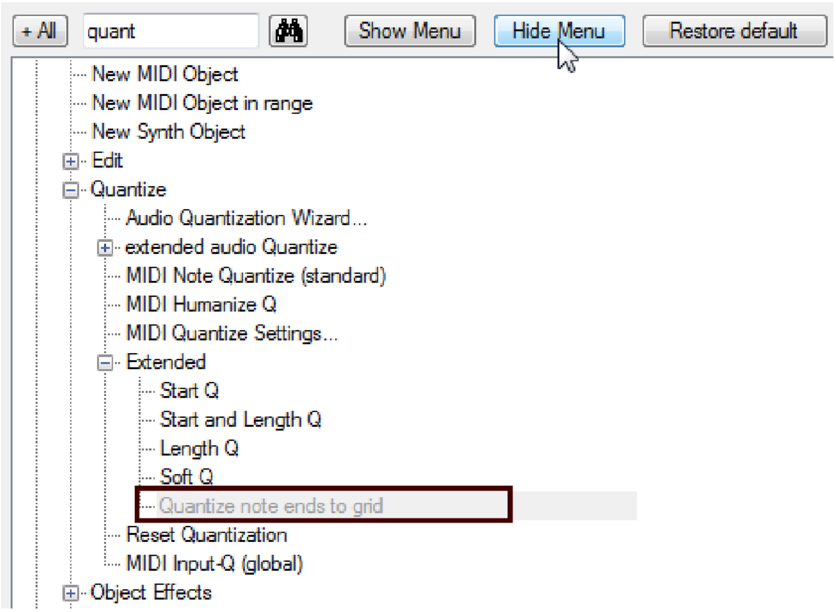
The main part of the dialog is the display of the complete Samplitude menu. Here you can select for which menu item you want to create a new shortcut or if the menu item should appear in the main menu of Samplitude.

Displaying and Searching for Menu Items

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



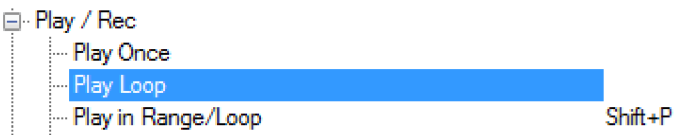
Show/hide menu item: Select a menu item which you would like to hide. "**Hide menu item**" removes the menu item from the menu It will be still shown in gray in the dialog's tree diagram.



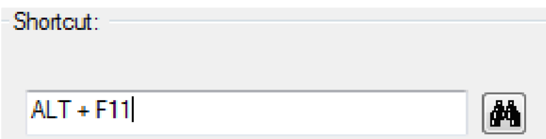
Please note that the menu item will no longer be selectable using the previously defined keyboard commands. "**Show menu item**" activates the hidden menu item again. "**Restore default**" resets the menu item to its default setting and makes the command visible again.

Creating shortcuts

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



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



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

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

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

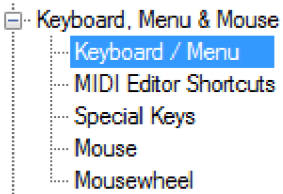
Deleting a shortcut

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

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

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

MIDI Editor/Special Keys/Mouse Wheel/Mouse



Under "**MIDI Editor Keys**" you can find or assign new keyboard shortcuts for editing MIDI events.

Sequoia only: With the **"Crossfade Editor keys"** you have numerous keyboard shortcuts available that you can use in the Sequoia Crossfade Editor.

"**Shortcuts**" redefines the keys for temporarily changing mouse and object modes. Use these functions to quickly switch from "Universal" mouse mode to a different mouse mode and to temporarily activate "Link objects" mode or to temporarily switch to "Write automation" mode, for example.

- Temporary switch for "Link one track (view page 115)": Activates the function "Link one track" as long as the assigned keyboard shortcut/key combination is held down.
- Temporary switch for "Link all tracks (view page 115)": Activates the function "Link all tracks" as long as the assigned keyboard shortcut/key combination is held down.

- Temporary switch for object contents: Enables object contents to be moved (view page 172) with the mouse as long as the assigned shortcut combination/key combination is held down.
- Temporary switch for "Snap (view page 99)" mode: Activates the "Snap" mode as long as the assigned keyboard shortcut/key combination is held down.
- Temporary switch for "Object" mode: Activates "Object" mode from "Range" mode as long as the assigned key is held down.
- Temporary switch for "Curve (view page 106)" mode: Activates "Draw automation" mode from "Universal/range" mode as long as the assigned keyboard shortcut/key combination is held down.
- Temporary switch 1 for Scrub Mode (view page 728): Activates Scrub Mode as long as the assigned keyboard shortcut/key combination is held down.
- Temporary switch 2 for Scrub Mode (view page 728): Activates Scrub Mode as long as the assigned keyboard shortcut/key combination is held down.
- Temporary switch for "Zoom Mode: Activates Zoom Mode as long as the assigned keyboard shortcut/key combination is held down.
- Switch for "Scrub Mode (view page 143): Switches to Scrub Mode indefinitely until the next time play stops.
- Temporary switch for "Cut Mode (view page 110): Activates Cut Mode as long as the assigned keyboard shortcut/key combination is held down.
- Multi-tap sequence key: Pressing this key multiple times provides special functions.
- Temporary switch for "Automation write (view page 440)" mode: Activates "Automation write" mode as long as the assigned keyboard shortcut/key combination is held down.

> More information about the different mouse and object modes is available in "Screen elements -> Program interface - overview".

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

- **Disable range zoom with double-click**
- **Zoom lasso allows vertical zoom without "Shift"**
- **Disable zoom with vertical mouse dragging on the timeline**
- **Downward-compatible use of "Shift"...**: This allows you to change the keyboard shortcut for "Switch to exclusive" for solo, mute, and record. Click "Solo/Mute/Record" and the selected shortcut "Shift + Alt" or "Shift" switches the individual channels temporarily to "Exclusive" mode.
- **Knob characteristics same as faders**: If this option is active, knobs can be pulled up and down like faders by dragging with the mouse.
- **2nd click required to move object**: This option requires a second click to moving objects. The first click selects objects and prevents unintentional moving.
- **Move thresh**: When an object is selected by clicking it, it could be moved unintentionally. If "Movement delay" is activated, Samplitude will wait a set time before executing the movement.

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

Note: To help Samplitude beginners switching to Samplitude we changed the mouse wheel behavior to match the standard behavior in many other DAW. With version Pro the mouse wheel scrolls in VIPs vertically (tracks) by default instead of horizontally (time position). With the „Reset“ button you can reset Samplitude to either the old or the new standard behavior.

Note: Pressing the key combination "Alt + Shift" might cause Windows to switch keyboard languages. In order to avoid this simply deactivate the keyboard shortcut used to switch input settings by going to the Control Panel and clicking on the "Clock, Language, and Region" icon. Navigate to the "Change keyboards or other input methods" tab and then click on the "Change keyboards" button to access the settings. In the "Advanced Key Settings" tab click on "Change Key Sequence" . Set the "Switch Input Language" and "Switch Keyboard Layout" options to "Not Assigned". This will prevent you from accidentally changing the input settings.

Edit toolbars

see Customize Toolbar (view page Fehler! Textmarke nicht definiert.)

Reset Toolbars

Resets the toolbars to default settings. Select whether individual toolbars should be reset or if all of them should be reset via "**Reset all toolbars**". If the file "Samplitude_TB.ini" is deleted from the Samplitude folder, then the default settings will be reset.

Project Display

Definition...

This dialog can be used to change between the two viewing options "**Draw mode 1**", "**Draw mode 2**", „**3**“ and „**4**“. Wave form display, object, VIP, and wave form color categories can be edited.

Keyboard shortcut: Shift + Tab

Mode 1/Mode 2/Switch Modes

Select a view mode here, or switchback and forth between the two viewing modes.

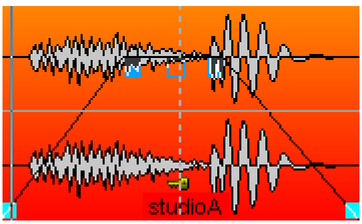
Shortcut: Tab

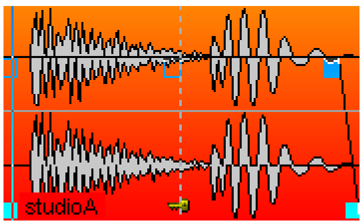
Waveform Display

Draw wave form: Turn the wave form display on and off here. When deactivated, you can see volume and panorama curves more clearly.

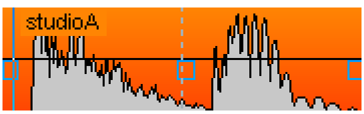
Note: In "Drawing mode 2", the standard default setting for the wave form display is switched off. This may be changed at any time.

Scale with fades/curves: In this default display mode, the wave form overview is scaled according to the settings for the fade in/out or volume curves. This makes the decay of the sound during a fade out or for volume curve changes more visible. This mode also offers good visibility when crossfading audio material.

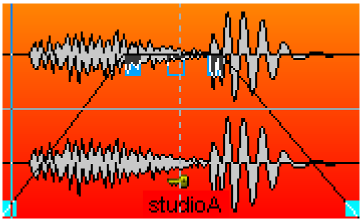




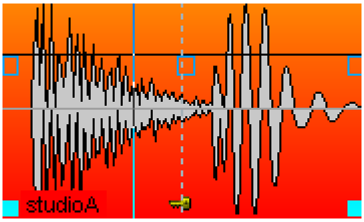
Halved waveform: Switches the graphical representation of the samples to half the waveform display size.

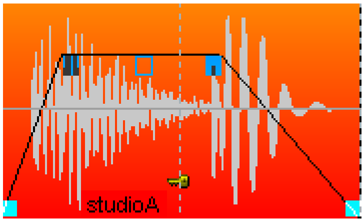


Separate stereo: If this option is active, stereo tracks are displayed separately one above the other.

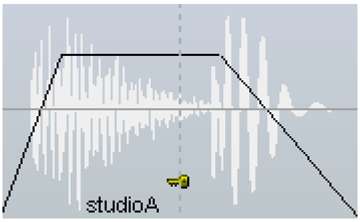


Draw envelope: If this option is active, an outline is drawn around the envelope of the waveform.



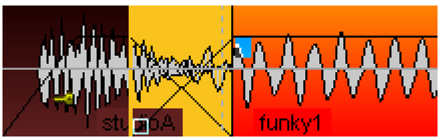


Gray out muted objects/tracks: Muted objects and tracks will be shown gray.

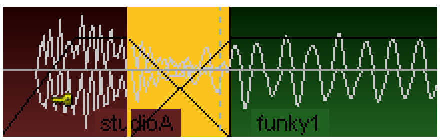


The second column is especially useful for a more clear crossfade display.

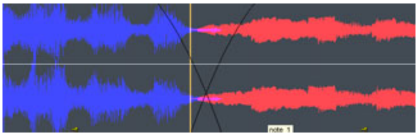
Standard: The waveform of the second object is drawn over the waveform of the first object:



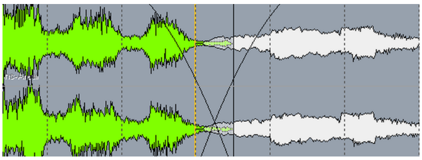
Envelope only: When this radio button is selected, only the waveform outline (envelope) is drawn. Both objects are well visible in crossfades:



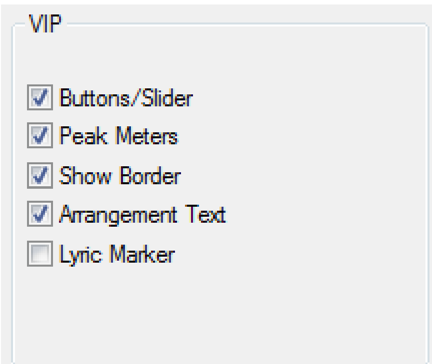
Transparent: If this button is selected, the colors of both crossfade waveforms are overlapped. The waveforms appear against a dark background. We recommend combining this and the "Waveform colors -> Red/blue alternation" to optimize the display of crossfade areas:



Interleaved: When this option is selected, a sample of the left object and a sample of the right object will be drawn. You may then visually assess the fade area, a task made especially easy if the adjacent objects have different colors:



VIP



Various VIP components (buttons/sliders, peak meters, show border and arrangement text) can be activated/deactivated here.

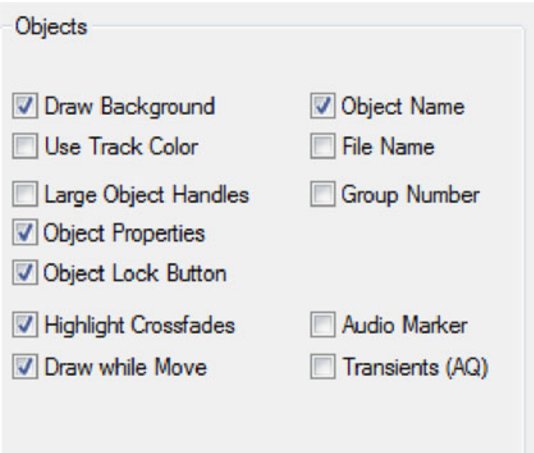
Buttons/slider displays the track header with all mixer controls, i.e. the channel faders, solo buttons, etc. If you always have the track editor open, you won't have to use these functionally identical control elements here. However, if the track header is left open, then only the track header's **peak meter** will be able to be hidden.

If you **hide the borders**, the edge of the selected track will disappear.

Arrangement text identifies the tracks for the two editing ranges in "Universal" mouse mode. If you set a check mark here, you will see "Play cursor and segment manipulation area" in the top track half, and "Object manipulation area" in the bottom half.

Lyrics marker activates the display of the lyrics markers.

Objects



Draw background: This activates the background color view of the selected objects. Every object can be assigned its own color (see "Object .> Object Color/Name (view page 709)" or in the Object Editor under "Color").

Use track color: The track color will be used as the background color.

Large object handles: The object handles are displayed larger.

Object properties: This option enables object settings like EQ, dynamics, panorama, effects or plug-ins to be displayed with abbreviations in the object.

Object lock symbol: If a checkmark is set, a key will appear on the bottom of an object, which can be used to lock and unlock objects (view page 692).

Highlight crossfades: With this setting, the crossfades between the objects are set in contrasting colors.

Draw while moving: The user interface is updated while moving the mouse. This option can be deactivated for computers with low processing capabilities.

Object name: Shows the object name.

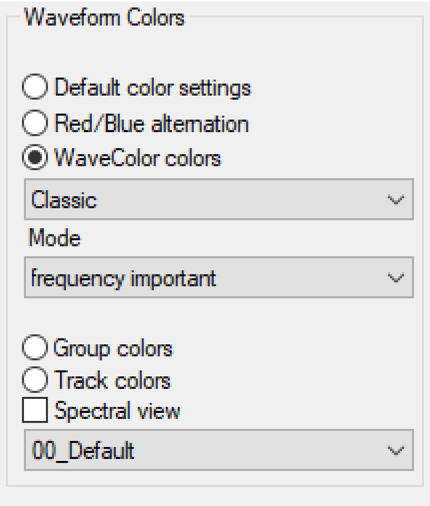
File name: The file names will be displayed.

Group number: Objects in a VIP can be grouped. The groups are numbered. When displaying these group numbers a quick overview appears showing which objects belong to which groups.

Audio Markers: Use this option to display markers that are present in the audio files in the corresponding objects.

Transients AQ: Use this option to display the transients in objects that were created beforehand with audio quantization.

Waveform color



Default color setup: The waveform contains the default color set in the color setup.

Details about the color setup can be found in the menu reference under "File > Program Preferences > System/Audio -> Design -> Colors (view page 614)".

Red/Blue alternation: This display mode colors the objects beside the waveform alternately red and blue so that crossfades in the "Transparent" and "Interlaced" draw modes are optimally visible. This way, crossfades can be optimally viewed in the "Transparent" and "Stacked" drawing modes.

WaveColor color gradient: With WaveColor, two additional properties of the audio material are visualized in the waveform display using color. The pitch is represented by the hue: Low tones are red, mid tones are green, high tones are blue. The sound characteristics of the audio material are represented by the color saturation: The more tonal and harmonic the signal, the more saturated the color, the more noise there is, the less saturated the color will be.

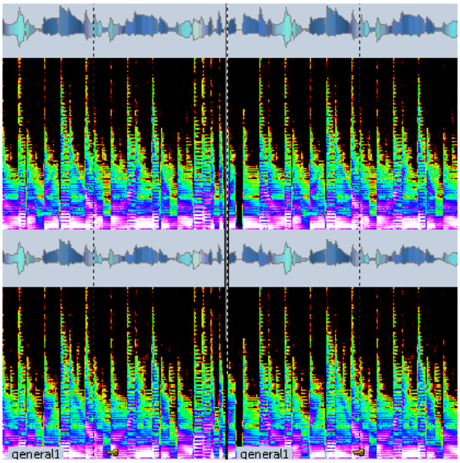
Various settings can be found in the drop-down menu: The "default" setting applies intense rainbow colors to the audio signal. The "Classic" presetting is similar to the colors used by Comparisonics. "Black-and-white pitch" and "Black-and-white noise factor" show only the pitch level or the noise components.

On the basis of the WaveColor colors, a completely new audio search function is available under "Object > WaveColor Audio Search (view page 719)" which can be used to easily find identical or similar audio material in your project.

Group colors: Object groups receive a randomly created individual group color when being made. This way, various groups can be displayed and differentiated very clearly from one another.

Track colors: The tracks will appear in colors selected under track options. The track options can be opened by right-clicking on a track name.

Spectral display: Here you can select various spectral color palettes. The graphical data that has been created will be saved in an extra *.hs file.



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

Color Settings

Details about the color setup can be found in the menu reference under "File > Program Preferences > System/Audio > Definition > Color Setup... (view page 614)".

Dithering Options...

How does dithering work?

An audio signal is quantized at a higher or lower resolution with every A/D conversion. During quantization, the signal receives an echelon form via limiting of the possible amplitude values. An 8-bit signal, for example, possesses only 256 amplitude values. This echelon formation causes distortion and thereby deforms the signal, especially impairing the sound of the lower levels.

Dithering can be understood as "mixing in" of low-level noise which seriously reduces the human ear's detection of sound defects.

When to dither?

Samplitude always dithers a signal in integer format when it is saved or exported.

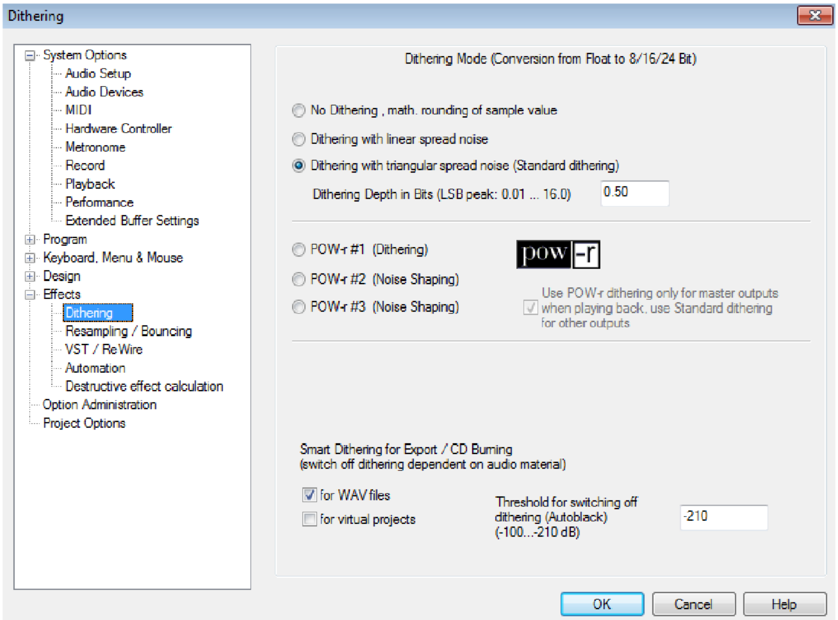
This is the case for the following situations:

- During 16-bit playback. All device addressing takes place in fixed comma resolution.
- When burning audio CDs in real time using an internal precision setting of 32-bit float.
- When using track-bouncing for virtual projects into 16-bit wave files, provided the internal precision is set to 32-bit float.
- When converting any 32/24-bit wave project into a 16-bit wave project.

Note: Samplitude does not execute dithering during recording.

The individual settings for dithering can be set during each trackbouncing process. Detailed information is provided in "Tools -> Trackbouncing (internal mixdown) -> Trackbouncing settings: options".

Dithering Options



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

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

Dithering with triangular spread noise (standard dithering): In this mode, audio data at 32-bit float is converted via dithering with noise featuring amplitude values split into triangular intervals. This means that values are more regular in the medium range and less often at the maximum or minimum values. This type of dithering usually creates better results than linear dithering. The noise is not modulated through the signal, resulting in a fading signal being enveloped by one constant noise signal.

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

POW-r Dithering/Smart Dithering

POW-r #1 (dithering): This function uses a special dithering curve to minimize quantization noise.

POW-r #2 (noise shaping): This function uses additional noise shaping across a wide frequency range to extend the dynamic range by 5-10 dB.

POW-r #3 (noise shaping): This function uses additional, optimized noise shaping to extend the dynamic range by up to 20 dB between 2 kHz and 4 kHz. The human ear is most sensitive to this frequency range.

> **Noise shaping** minimizes side effects caused by bit reduction by spectrally displacing the quantization noise above 10 kHz, which is the range the human ear is least sensitive to.

Which dithering mode sounds the best depends mainly on the audio signal.

Use POW-r dithering only for master outputs when playing back, use standard dithering for other outputs: If this option is selected, then only the master outputs will be dithered with the POW-r dithering algorithm. The individual outputs will use standard dithering, i.e. **dithering with triangular spread noise**.

Smart dithering for export/CD burning (switch off dithering dependent on audio material)

For wave files: This option is set as the default. When burning CDs and exporting 16-bit files, dithering will only take place when the file is not currently 16-bit. Dithering is not calculated during silence.

For virtual projects: This option is switched off by default. In this case, dithering only occurs if the bit depth does not equal 16-bit.

Threshold for switching off dithering (autoblack): Specify a threshold value below which any dithering noise that is created should be muted. The range of values in this case is between -100 dB and -210 dB.

Resampling / Freeze Options

Use these options to set the following resampling/freeze calculations.

Resampling options

- Record resampling (chase lock sync, conversion to 44.1 kHz during recording)

- Playback resampling (scrubbing, chase lock sync, playback at altered sampling rate)
- Object resampling for new objects (load objects with sample rate different from project)
- Resampling during bouncing (prior to burning CD)

Time Stretching/Pitch shifting options

Here you can set the standard algorithm (view page 808) for objects.

Freeze options

- Keep mono, if possible
- Use additional samples for object freeze
- Object freeze without object volume
- Format (16-bit, 24-bit, 32-bit)

Set Preroll Time...

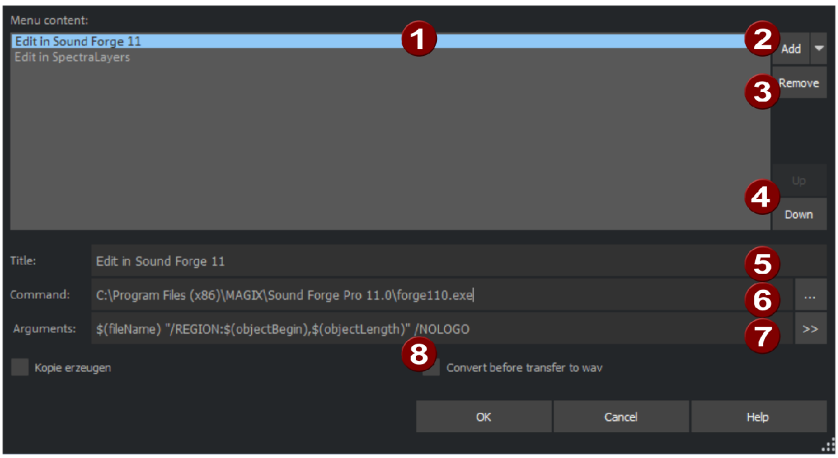
This option sets the pre-roll time for the cut simulation functions. The cut simulation function is started via the corresponding buttons in the punch/playback bar. The pre-roll time specifies how long is played until the selected range is reached.

Destructive Effect Calculation...

Choose in the advanced destructive effects calculation options (view page 757) if you want to add the effect to the original file when editing destructively, write it into the effect file, or create a new effect file for each calculation.

External Tools

This dialog is where you can configure objects for editing other audio programs.



1. **Menu content**: All entries in this list create entries in the context menu for an object (view page 173). The entries can be transferred to an external editor for editing.
2. **Add**: Click here to add a new tool. To do so, search for the executable file you need in your program folder. If Sound Forge or SpectraLayers is installed, the tool may be entered automatically with all its parameters via the dropdown menu.
3. **Delete:** Deletes the selected entry. If you delete an entry and close the dialog that then appears with "OK", the entry will no longer be displayed in the context menu for objects.
4. **Up/Down**: Use to change the sequence of menu entries.
5. **Title:** Enter text for the menu entry here.
6. **File selection dialog:** Search for an executable file on your hard drive using the file section dialog.
7. Predefined placeholders: In this menu you'll find the predefined placeholders to set in the commando line in order to transfer object parameters e.g. object start and end to the external program, if it supports them.
8. **Convert to wave before transfer:**Since some editors can't open MP3, FLAC or video files, these types of files are converted by the external editor and saved under another name. However, this will prevent you from transferring the file back to the virtual project. We recommend that you activate this option when working with these editors. This means the content of an object will always be converted to wave before transferring to the editor and the object will be adjusted for editing.

Option Administration

See „System Options“ > „Option Administration“ (view page 618).

More

Internet Connection

This menu item allows you to terminate an existing Internet connection directly from within Samplitude.

Append Project

With this function, a project can be appended to another one. The objects or wave files of a selected project will be immediately copied to the end of an existing object.

To execute this command, first click the project you would like to attach the material to. Next, select the "Append project" command, followed by a mouse click the project to add to. Finally, confirm via the appearing dialog by clicking on the "Add" button.

Add a VIP to a VIP

Virtual projects may be inserted into other virtual projects and added to the end of already existing virtual projects.

All objects of the appended VIP remain separated and are inserted at the end of the existing project from the first track downwards.

Note: The mixer settings for the inserted project will not be applied with this command.

Add Wave Project to a Wave Project

"Append Project" allows you to copy a wave project to the end of an already existing wave project.

Recent File

In the last section of this menu the 10 projects opened last will be displayed.

Edit Menu

This menu features functions that are applicable only to wave projects (e. g. „Copy “ > „Copy as“), and others that are only for VIP projects (e. g. „Insert Silence“).

Edit functions like "Delete > Extract" require a selected range; other functions like "Copy" are available only for active objects. The start and end points of the selected range determine the start and end points of a cut, and the vertical position indicates the tracks that are affected by the edit.

Activate the auto crossfade function and all the newly created fade edges will automatically be made soft or disappear.

Mouse mode

The following options allow you to select from a variety of mouse modes like in the Mouse mode (view page 102) selector on the upper tool bar.

Universal Mode

This is the preset mouse mode for Samplitude. All necessary functions are available by left-clicking. Right-clicking opens a context menu.

> Detailed information about Universal Mode is available in "Screen Elements > Toolbar - Overview > Mouse Mode > Universal Mode".

Range Mode

In this mode, only ranges and the playback marker can be manipulated.

> Detailed information about Range Mode is available in "Screen Elements > Toolbar - Overview > Mouse Mode > Range Mode".

Curve Mode

This mode allows panorama and volume curves to be drawn, edited and deleted with the left mouse button.

> Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Curve Mode (view page 106)".

Object Mode

In this mode objects can be moved and curves can be edited with the left mouse button.

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Object Mode (view page 107)".

Object/Curve Mode

In this mode, you can move objects and edit curves using the left mouse button.

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Object/Curve Mode (view page 108)".

Left/Right Mode

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

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Left/Right Mode (view page 109)".

Wave (HDP/RAP) Mouse Mode

Range

Use the left mouse button to set the playback marker and select ranges.

Draw Wave

Draw the sample material at a high zoom level, which can be helpful if you want to remove clicks and clipping.

Volume Draw Mode

In this mode, you can directly manipulate the volume using the mouse.

Scrub Mode

This mouse mode enables you to preview and control the playback speed. The project is played forwards or backwards, depending on the scrub direction.

Zoom Mode

Click with the left mouse button to zoom into the wave display; zoom out with the right mouse button.

Cut Mode

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Cut Mode (view page 110)".

Pitch Shift / Time Stretch Mode

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Pitch Shift / Time Stretch Mode (view page 110)".

Volume draw mode

This mode allows Volume Curves to be drawn with the left mouse button. You must first activate the volume curve on the corresponding track with the "Vol" button to the left in the track header. Then, click on the curve to create curve points. These can then be dragged to the desired position.

Left mouse button:	Freehand drawing function for volume curve.
Right mouse button:	Context menu

Automation Draw Mode

This mode allows automation curves to be drawn by using the left mouse button. Activate the automated parameters for the embedded plug-in beforehand by pressing "**Ctrl + Alt + Moving the corresponding plug-in control element**". Next, click with the right mouse button on the "Track automation" field in the track editor, and place a check mark for the desired parameters. Now you can draw in the curve of the selected parameter in the track in Automation Draw mode.

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Automation Draw Mode (view page 111, view page 442)".

Spectral Mode

In "Spectral Mode (view page 111)" you can remove noise from a track object without the wanted signal being audibly influenced. Editing can be performed directly in the arranger window.

After switching to "Spectral Mode", select the noise by clicking and dragging out a rectangle over it. The handle points can be used to make fine adjustments to the selected area. After making the correction, you can see the results in the Wave/Spectral display in the arranger window.

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Spectral Mode". (view page 111)

Preview (Scrubbing Mode)

This mouse mode enables you to preview and control the playback speed. The project is played forwards or backwards, depending on the scrub direction.

Zoom Mode

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Zoom Mode". (view page 112)

Color Mode

Detailed information about this can be found in the chapter "Screen Elements > Toolbar - Overview > Mouse Mode > Color Mode". (view page 113)

Object mode

Detailed information about object modes can be found under Symbol bars > Object modes" (view page 114).

Undo

Samplitude offers a secure way of tracking your changes in Virtual and Wave Projects, which makes it easier to undo work steps. Up to 100 changes can be kept in the memory using the undo function in "Menu > File > Program Preferences > Undo Definitions (view page 624)".

When editing audio material with destructive effects ("Offline Effects menu"), the Undo option only works when the option "Create copy" is activated in each dialog.

Shortcut: Ctrl + Z

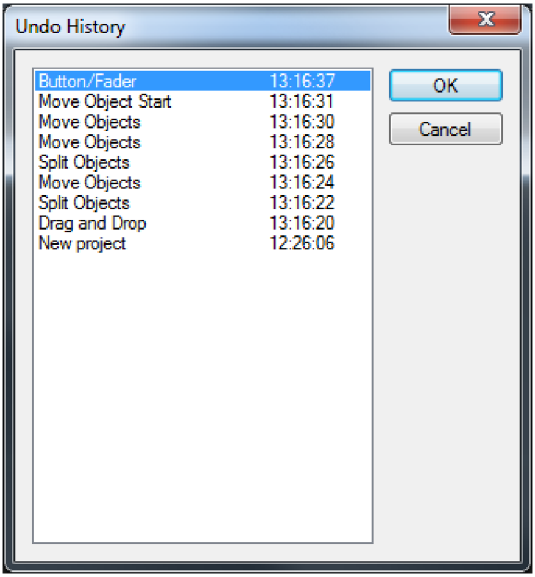
Restore

"Redo" (undoes) the most recent Undo command.

Shortcut: Ctrl + Y

Undo History

A list featuring the most recent executed commands appears.



In this dialog, you can return to an earlier work stage. This list can be removed in the "Edit -> Delete undo levels (view page 648)".

Delete Undo History

Deletes the undo levels of a project.

Copy

Copy

The current range is copied into the clipboard. Please note that any previous clipboard content will be deleted.

Shortcut: C, Ctrl + C or Ctrl + Ins

Copy as... (Wave-Project)

This function copies the selected range from a wave project into a new file.

Shortcut: Shift + C

Copy + Clear

Copies the current range to the clipboard (clip), removes it from the project, and replaces it with silence.

Shortcut: Ctrl + Alt + C

Cut

Cut

Virtual Projects (VIP)

In the case of virtual projects, the selected area will be copied into the virtual clip (VirtClip) and removed from the active project. The VirtClip, like the virtual projects, does not contain the actual audio files, just the links to them. The VirtClip spans over the same number of tracks as the selected area.

Keyboard Shortcut: X

Wave projects

The data of the selected range is copied into the clip and removed from the current project. Data behind that range takes its place. The Wave project is reduced in size by the length of the removed segment.

If you cut a piece out of a mono wave project, the clip will be a mono wave project, and if you cut out a piece from a stereo wave project, the clip will also be a stereo wave project. Similarly, the bit resolution and the sample rate from the project are applied. The old content of the clip is overwritten.

After the function has been carried out, the playback marker will be at the cut edge of the removed segment. You can now place the clip back into the wave project with the command "Paste from clip".

Cut with Time/Ripple

This command deletes the marked range and copies it into a clip. The material behind the removed range will be moved forward by the length of the range (ripple, shuffle).

Shortcut: Ctrl + Alt + X

Split

Split Objects

Use this option to split all selected objects at the cursor position and separate them into individual objects.

If the range is selected, the cut will be made at each border. Ensure that the object that is to be edited is contained within the active range. If necessary, all objects may be activated by double clicking on the range selected in the range bar. In this case, it is not necessary to select the objects beforehand.

Shortcut: T

Split Objects at Project Markers

The object is split into separate objects at the marker positions within the object borders. The newly created objects will be named after the previous markers.

Split Objects at CD Track Indices

The object is split into separate objects at the marker positions within the object borders. The newly created objects will be named after the previous track markers.

Split Objects at Audio Markers

The object is split into separate objects at the audio marker positions within the object borders. The newly created objects will be named after the previous audio markers.

Heal/Unsplit Objects

This command recombines two objects that have been split without any audio calculation. To do this, the objects in the VIP must be positioned exactly beside each another, and the audio material must be connected. The objects must therefore be different but sequential pieces of audio material from the same wave project. If this not the case, the command will not function properly.

Before executing the command, select both of the objects that were split previously. All of the second object's properties, like fades and effects, will be discarded. The resulting combined object will feature the properties of the first object.

Insert

Paste/Insert Clip

The data contained in the clip or VirtClip is inserted at the current position of the playback marker (position line) or at the beginning of the current range.

In the "Link objects" modes, objects behind the insert point are moved to the end. In all other modes, the inserted material replaces the existing objects.

A new area will automatically be defined for the inserted files.

Shortcut: V, Ctrl + V, Shift + Insert

Paste with Time/Ripple

This menu option inserts the clip at the position of the cursor or at the start of the range. The audio material after the cursor position will be moved down by the length of the range (ripple, shuffle).

Shortcut: Ctrl + Alt + V

Overwrite with Clip

The current range will be overwritten with the contents of the clip. The length of the clip will not be reduced. The data previously located at this position will be overwritten.

In virtual projects, the selected range determines the start position and track for the clip contents.

Shortcut: Alt + V or Insert

Mix with Clip

The content of the range is mixed with the content of the clip. This command is available only in the destructive wave edit mode. Please note that with this function both components are entered into the mix at 100%, being added together. This can cause overmodulation. You may have to adjust the amplitude of the project by normalizing prior to this.

Crossfade with Clip

The content of the range is crossfaded with the content of the clip. This command is only available in the destructive wave edit mode. In this case, the position of the playback marker specifies the end of the clip to be faded with the wave project. The length of the clip sets the length of the crossfade.

Delete

Virtual projects (VIP)

Use this command to delete the selected objects from the virtual project. If a range is selected, then the track automation curves will also be deleted in the range. If objects are only partially in the range, then the corresponding parts of the objects will be deleted.

In connection with the object modes (view page 114), "connect objects..." moves the objects located behind the deleted range forward, and the track will be shortened accordingly. In the other object modes, the range is replaced by silence.

Keyboard shortcut: Del

Deleting Audio Files

The data in the selected range is deleted and the subsequent audio segment is moved back to fill the gap.

> Caution: If you are using offline Audio Edit Mode and have deactivated the "Undo" function ("Options -> Program preferences -> Undo Definitions (view page 624)"), the files cannot be restored.

Ripple delete

This command deletes the marked range. Material later than the end of the range will be prolonged by the length of the range.

Shortcut: Ctrl + Del

Extract

Press "Extract" to trim your project by deleting the areas outside of the selected segment.

Virtual projects (VIP)

The extraction function results in all objects outside of the selected segment in a virtual project being deleted. All tracks stay unchanged, even if no object is found in the selected segment. This function is not track-specific.

Individual objects can be cropped using "Object -> Trim Objects (view page 691)".

Extracting Audio Files

In Wave Editing Mode only the part of the audio file below the selected area remains. The data before and after the range is deleted.

Silence

Insert Silence

This command inserts silence to all tracks at the position of the current play or at the start position of the selected range. The objects will be split at this position and the files after it will be added after the silence.

If a range is selected, its length will be adapted as default. The selected range is not adjusted after the operation. It's possible to change the value in the input field. The project is extended by the length of the inserted object.

Strip silence

Use this function to split objects at positions where silence occurs. After applying the command the objects are split in such a way that the silent parts of the objects are separated from them. You can specify a threshold value for both silence and wanted signal in the dialog beforehand. The cut objects are selected to make it easier to delete them using the "Del" button.

Threshold for silence detection (dB): If the signal falls below this value, the object will be cut at the corresponding position.

Threshold value for used signal (dB): If the signal rises above this value, the object will be cut at the corresponding position.

The "**Minimum length of silence**" parameter sets the length which silent passages should be applied so that they can be selected and deleted. This way, you can remove very short passages of silences from the cut.

With the "**Crossfades**" option, you can create automatic crossfades between at the cut positions of the objects.

Use sample space accuracy: With a new, improved algorithm, the processing speed of this function can be significantly increased. If absolute accuracy is not important when cutting, disable the option for a speed advantage.

Replace with silence

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

Keyboard shortcut:	Alt+Del

Tempo

Tempo map...

Detailed information about the tempo map dialog can be found in the chapter "Tempo editing > Tempo map dialog" (view page 399).

Tempo map BPM mode/grid adjustment mode

Detailed information about the tempo map grid fit mode can be found in the chapter "Tempo editing > Grid tapping" (view page 402).

Insert tempo change

This opens the Tempo and time signature dialog (view page 407) to insert a tempo change.

Insert time signature change

This opens the Tempo and time signature dialog (view page 407) to insert a time signature change.

Insert grid position marker

Here you can insert grid position markers (view page 398) at the current playback marker position.

Change Global Tempo

In this dialog you can change the tempo of the project by entering a new value. In this case the value represents the change factor whereby "2" means that the tempo is doubled (in BPM) and "0.5" means that the tempo is halved.

Ignore all tempo markers, use project tempo only

With this option you can ignore any previously created tempo markers in your project, so that only the project tempo observed.

Metronome Active

This command activates/deactivates the metronome. This function can also be activated via the "Click" button in the transport controls.

Metronome options

> Detailed information about this is available in "File -> Program Preferences -> Metronome Options (view page 44)".

Create click track

The new command **Menu Edit > Tempo > Create click track** creates an audio track, which contains all metronome clicks as objects.

Range

Range All

This expands the range to include the entire project. Double clicking the timeline or pressing "A" selects the range in the selected track; double clicking again or pressing "A" again selects the range over all tracks, and double clicking or "A" once again restores the simple timeline selection.

Shortcut: A

Manipulate Range

Move Range Start Left

This function moves the beginning of the range in the current arranger window one grid unit to the left. The length of the movement depends on the set grid value. This may be adjusted via "Project options -> General" (shortcut: "I", "Ctrl + Shift + #").

Shortcut: Alt + ÷ (number block), right arrow

Move range start to right

This function moves the beginning of the range in the current arranger window by one grid unit to the right. The step size is defined by the set grid value.

Keyboard shortcut: Alt + * (number pad), right arrow

Move range end to left

This function moves the beginning of the range in the current arranger window one grid unit to the left. The step size is defined by the set grid value.

Keyboard shortcut: Alt + - (number pad), Shift + left arrow"

Move range end to right

This function moves the end of the range in the current arranger window one grid unit to the left. The step size is defined by the set grid value.

Keyboard shortcut: Alt + + (number pad), Shift + right arrow

Range to Beginning

Choose this option to extend the currently marked range to the beginning of the project (Wave Project or VIP).

Range to End

Choose this option to extend the currently marked range to the end of the project (Wave Project or VIP).

Flip Range Left

Choose this option to flip the currently marked range to the left (this swaps the start of the marked range to the end of the marked range.)

Shortcut: Ctrl + Shift + left arrow

Flip Range Right

Choose this option to flip the currently marked Range to the right. (What used to be the end of the marked Range is now the beginning of the marked Range.)

Shortcut: Ctrl + Shift + right arrow

Find zero crossings

Beginning of Range -> 0

Choose this option to move the beginning of the currently selected range to the next zero amplitude crossing. "Zero" is the next sample value featuring a ZERO value or that indicates the boundary between a positive and a negative sample value (or vice versa). This function is especially useful when searching for loop points. Remember to zoom into the waveform display far enough so that you'll be able to see the actual changes in the arranger window.

Shortcut: Ctrl + Page up

Beginning of Range <- 0

Choose this option to move the beginning of the currently marked range left to the previous zero amplitude crossing.

Shortcut: Shift + Page up

End of Range -> 0

Choose this option to move the end of the currently selected range to the next zero amplitude crossing.

Shortcut: Ctrl + Page down

End of Range <- 0

Choose this option to move the end of the range left to the previous zero amplitude crossing.

Shortcut: Shift + Page down

0 -> Range <- 0

Choose this option to move the beginning of the range right to the next zero amplitude crossing, and to move the end of the range left to the previous left zero amplitude crossing.

Range start to left marker

Choose this option to extend the Range to the next marker on the left.

Shortcut: Shift + F2

Note: This command also takes set object audio markers into account as soon as they have been selected via menu "File > System preferences > System/audio -> Design -> View options".

Range Start to Left Object Border

The start of the range is set to the next object edge on the left.

Range end to right marker

The end of the range is set to the next marker to the right.

Shortcut: Shift + F3

Note: This command also takes set object audio markers into account as soon as they have been selected via menu "File > System preferences > System/audio -> Design -> View options".

Range End to Right Object Border

The end of the range is set to the next object edge on the right.

Area Above all Selected Items

Choose this option to extend the range over all selected objects.

Remove Range

This command removes the current range from the grid and marker bar. This function is also available by right clicking in the grid and marker bar.

Range Not Over Tracks

This command selects the selected range from the grid and marker bar only, and not in the tracks. Switch between the "Range not over tracks", "Range over current track", and "Range over all tracks" functions by double clicking in the selected range in the grid and marker bar.

Range Over Current Track

This command selects the selected range from the grid and marker bar as well as in the selected track.

Range Over All Tracks

This command selects the selected range from the grid and marker bar as well as in all selected tracks.

Range Length

Set the range starting from the playback marker to lengths of 1, 2, 4, 8 or 16 beats here.

Split Range

Use this function to split the arranger window into three sections. In this case, the upper section displays the entire project in an overview, while the left lower section displays the start of the range and the right lower section displays the end of the range (enlarged sections).

The range borders may be set precisely in the lower sections, provided the grid setting "Grid/frames" is set in the "Project options" beforehand (shortcut: "I").

Shortcut: B

Back to range: "Shift + B"

Split Range for Video

This function is especially useful when working with AVI videos. The upper section displays the entire project, the lower left section displays the range start, and the lower right section indicates the range end. For the lower sections, a zoom depth of 1 frame is preset to make image-precise cutting and editing possible.

To undo the split, restore the original range by pressing "Shift + B".

Store Range

Samplitude enables ranges to be stored by selecting the corresponding range number.

All current ranges in a project may be viewed via "Tools -> Manager -> Range manager".

Shortcut: Alt + F2 ... F10

"Alt + F4" however, should not be used, since this is a Windows command that closes the current screen. Similarly, "Alt + F9" should not be used either, since this is used for the 4-point cut command in Sequoia.

The "Other..." dialog may be used to define additional ranges featuring freely assigned names.

Shortcut: Alt + F11

Get Range

Open previously saved ranges via this command. Getting a range is also possible during playback. This enables you toggle between various ranges in order to compare them acoustically.

Shortcut: Ctrl + F2 ... F10

Ranges may be named and accessed via "Tools -> Manager -> Range manager".

Get Range Length

Use this function to set the current range to the length of the corresponding saved range.

Shortcut: Ctrl + Shift + F2 ... F10

Range Manager...

The "Range manager" displays all saved ranges of the current project.

Detailed information about the "Range manager" is available in the chapter "Manager -> Range manager (view page 188)".

Shortcut: Ctrl + Alt + Shift + B

Recall Last Range

Use this function to recall the last selected range. Repeated application accesses ranges up to five levels back.

Shortcut: Shift + Backspace

Crossfade

Crossfade editor

This function has been revised in Samplitude. For the latest information, see the PDF document **Samplitude Pro X7 New Features** in the program folder.

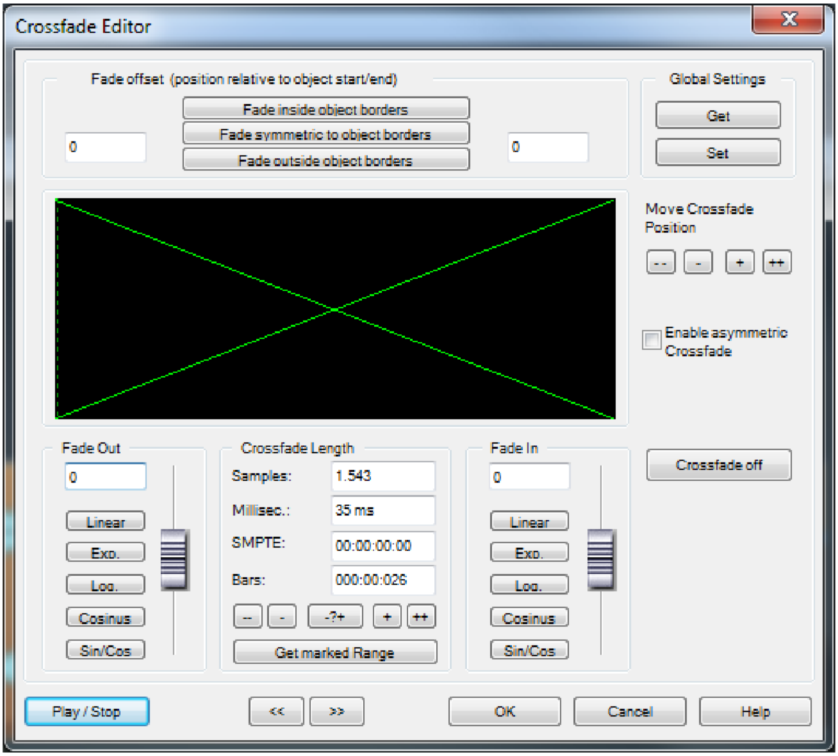
Samplitude features a professional crossfade editor.

The crossfade editor is opened by clicking the toolbar button or via menu "Edit -> Crossfade Editor". You can also use the keyboard shortcut "Shift + F".

Use the crossfade editor to conveniently and efficiently edit all aspects of a crossfade. All changes are non-destructive and can be undone.

Crossfading with the crossfade editor

In the VIP, select the objects which you wish to fade into and open the crossfade editor.



The following editing options are available for selection:

Crossfade length: The length of the crossfade may be set in different sized steps. Clicking the "+" / "-" buttons increases/decreases the length in small steps, the "++" / "--" buttons in larger steps. Use the "-?+" button to customize these step effects as needed. The crossfade length can also be defined by a selected range in the VIP by clicking on the "Get range length" button in the "Range" menu. This range has to be selected before opening the crossfade editor.

Fade in/fade out: There are plenty of different curve types available for selection for the crossfade:

- Linear (0)
- Exponential
- Logarithmic
- Cosine
- Sine/cosine

The faders may still be used to change the curves in the corresponding available range. Linear curves require less computer power than non-linear.

Note: The settings for fading in/out in the crossfade editor are applied to the selected object if it does not overlap any other object.

Global settings: Set

The current settings are defined as by default. This is useful in case an object has been split with the "T" key and you want to create a crossfade between the two objects. When opening the crossfade editor, the specified settings are set as default.

Global settings: Get

Use this button to get the presets of the crossfade editor.

Get range length: The range length of the range selected in the VIP is set as the crossfade length.

Fade offset: Specify how much of the fade should be outside of the object border. Usually, the entire fade is located within the borders of the object, that is, the fade starting point is at the start of the object (0%) as is normal with fade ins/outs of individual objects. If the fade becomes part of a crossfade, it may be necessary to modify it. For example, if the second object starts with a drum beat, it would be better to execute the crossfade beforehand in order to keep the "attack" of the drum beat.

The value may be changed from 0% (fade within the object borders) to 50% (fades symmetrical to the object borders) to 100% (fade outside the object borders) – the object is then stretched by the percentage of the fade length. The "actual" beginning of the object is displayed as a dashed line.

With a fade offset of over 0%, make sure that the corresponding audio material is available in the wave project, otherwise it may be the case that an object with a modified fade offset cannot be faded in or out. For example, if the object starts at the exact beginning of the wave project, you will no longer be able to crossfade if the fade offset rises above 0%.

Crossfade off: The "Crossfade off" button deactivates the crossfade of the selected object. The fade out time of the first object is set to 0.

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

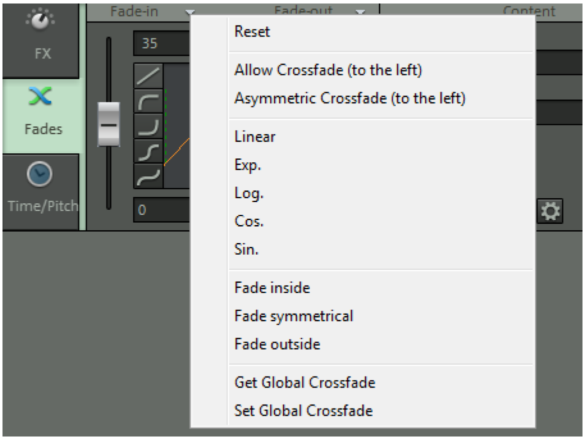
Move crossfade position: The end of the first object and the end of the second object are moved equally. The ++/--/+/- buttons are used to set the crossfade length.

Enable asymmetrical crossfades: The fade offset settings of both objects are no longer connected to one another and can therefore be set independently of one another.

Auto crossfade mode active

With this option, all newly recorded or cut material in a VIP has an automatic fade-in applied to the start and end.

Every object may be assigned a standard fade-in, and this may be edited via the object editor fade menu via "Get/set global crossfade".



If two objects overlap in this mode, a real-time crossfade will occur at the intersection. The fade settings can be modified using a double click on each object in the object editor.

If required, you can manually edit each intersection with the crossfade editor or object handles.

Batch Processing

Batch processing lets you automate work processes for several files. This makes it possible to define an editing sequence that will be applied to all selected files.

Each "job" is added to the job list. You can define multiple batch jobs which will then be executed one after the other.

Possible editing options:

- Normalization
- Linear fading (in and out)

- All of the realtime effects included in Samplitude
- Time Stretching / Pitch Shifting
- Remove DC offset
- Target format settings: Word width (8/16/24/32 float), sample rate, stereo/mono settings, file format conversion, compression process
- Save settings and name the created file while retaining the original

For example, you can normalize a whole folder full of 24-bit wave files to 96%, apply 5 ms fades at the beginning and the end of each file, compress them with the multi-band compressor, change them all to 16-bit mono and then save them all as MP3 files.

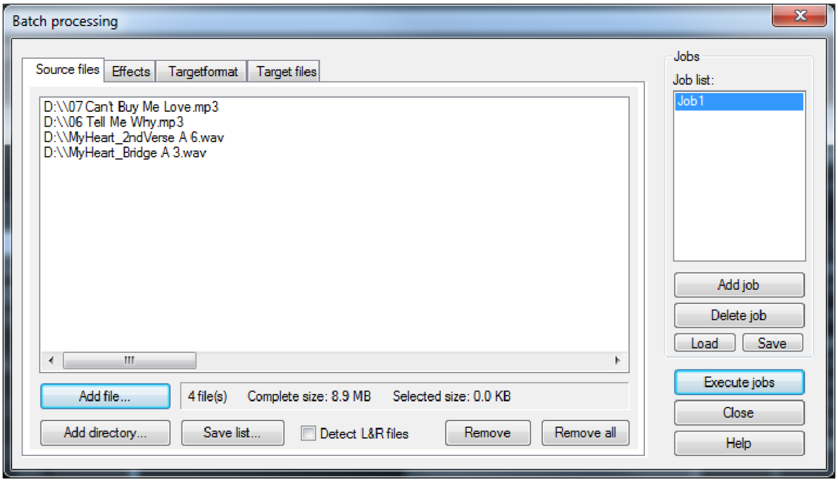
Jobs

On the right border of the batch processing window is the job list. New jobs are created by clicking the “**Add job**” button. If you select a job (with a mouse click), the four settings tabs are displayed (source files, effects, target format, target files). "**Delete job**" removes the selected job. “**Perform jobs**” starts the batch processing.

With **„Load“** and **„Save“** you can load and save jobs in XML format.

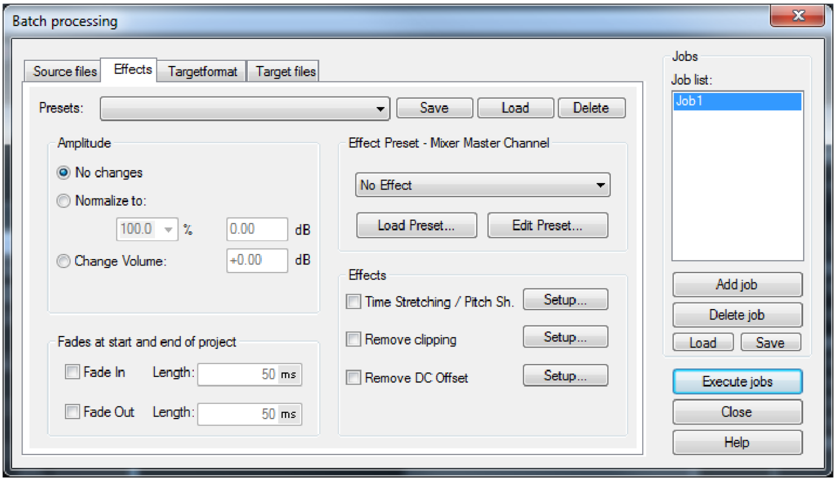
Even if you close the batch processing window temporarily, all created jobs will remain saved until Samplitude is closed.

Source Files



Create the list of the files for editing with the "**Add file**" button. Multiple selection is possible. Select **“Load folder”** to add all of the audio files in a folder (including all subfolders). You can load all file formats that Samplitude imports. "**Save list**" creates a playlist in *.m3u format for selection later. The option "**Detect L&R files**" makes it possible to edit left and right files as stereo. "**Remove**" deletes all selected list entries. "**Remove all**" deletes the complete list.

Effects



1. Normalize amplitude: You can enter a target maximum amplitude value in % or dB. For example, a value of 75% corresponds to -2.5 dB. You can also enter volume changes in dB.

Detailed information about normalization is available in "Effects > Amplitude > Normalize (view page 759)".

Loudness adjustment (only Sequoia): Here you can adjust the program settings for loudness in LUFS (e. g. EBU R128 Loudness Norm) for calculating loudness values. This way you can set the loudness value for selected files.

2. Load effect presets from the master channel: It is possible to integrate existing master channel effect settings into the batch processing using mixer-preset files.

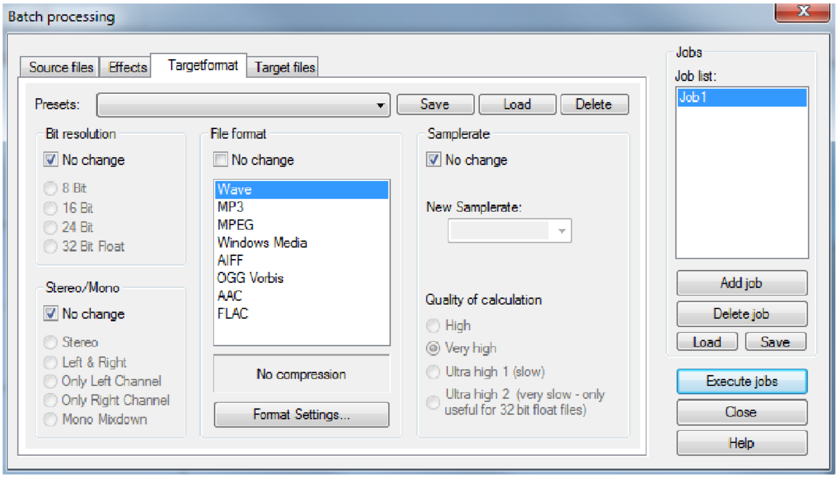
- The Samplitude mixer enables all of the mixer settings to be exported as a mixer preset. To do this, click the "Save mixer settings" button in the mixer. This is located under the mixer snapshot storage slots.
- When you load the *.mix file into the batch processing function with "**Load preset**", all effect settings for the master channel will be calculated into the file batch. The "**Edit preset**" option opens a special FX routing dialog that provides access to all of the effect parameters.

Detailed information about the FX routing dialog is provided in the chapter "Mixer - buses and routing -> Effects routing/plug-ins dialog (view page 239)".

Additional effects: In addition to the master effects of the mixer presets, non-real-time effects such as "Pitch Shifting/Time Stretching", "Remove DC Offset", and "Remove clipping" can also be applied. The "Settings" button opens the corresponding effects dialog.

3. Fades at start and end of project: Linear fades can be added to the start and end of files. The fade lengths can be set freely.

Target format

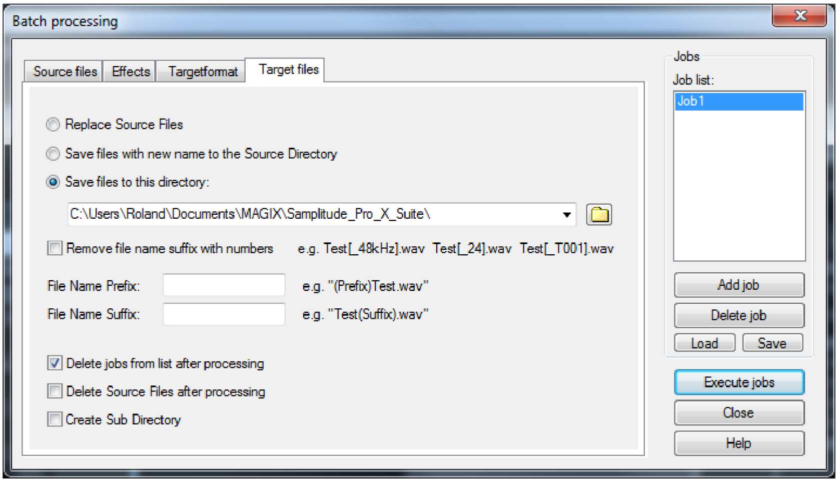


All settings in this dialog can be saved as presets. You can freely choose the bit resolution (8/16/24/32 bit float), the sample rate, stereo/mono/left/right as well as the save format with the corresponding format settings.

You can find out more about sample rates in the "File > Effects > Sample manipulation > Adjust sample Rate (view page 848)".

You can find out more about file formats in the "Menu > File > Export (view page 575)".

Destination Files



There are various ways to save edited files:

Replace source files: The original file is replaced with the edited one. If the file is used in a VIP, the VIP is closed first.

Save files in the source folder with a different name: The changed file will be copied back into the source folder and the file name will be expanded with the added "**file name prefix**" and "**file name suffix**".

Save files in the following source folder: The changed file will be copied back into folder that can be freely selected and the file name will be expanded with the added "**file name prefix**" and "**file name suffix**".

Remove file name suffix with numbers: This option removes numbers added by the Samplitude batch processing function (e. g. "_48kHz" or "_T001").

Delete jobs from list after processing: If you select this option, a job that has been processed will be removed from the list.

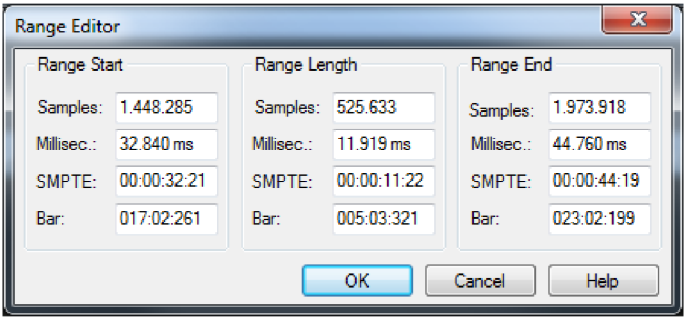
"**Delete source files after processing**" can also be applied.

Keep source folder structure: This option saves all files in the same subfolder structure including the source path. This process creates subfolders as required.

More

Range Editor...

This window specifies the currently marked range's start and end positions and length in various measurement units.



Adjustments to the parameters have the following effect:

Change start of range -> Range end remains constant

Change range length -> Range start remains constant

Change range end -> Range start remains constant

Edit time display

Here you can set and the number fields for the range position (Alt + Num pad 1), range length (Alt + Num pad 2), and range end. (Alt + Num pad 3).

Shortcut: Alt + Num pad 1 ... 5

Track Menu

Insert New Tracks

Add One Track

Use this function to add a new track to the last track in the project.

Add Several Tracks

To add multiple tracks, select the command "Add several tracks". A dialog window appears to select the desired number of new tracks.

The maximum track count in Samplitude is limited to 999 stereo tracks. Each of these tracks can also act as a AUX bus and/or as a submix bus.

Insert Track(s)

This command adds one empty track at the end of a selected track.

Insert several empty track(s)...

To add multiple tracks, select the command "Insert several empty tracks". A dialog window appears to select the desired number of new tracks.

New MIDI Track

This command adds a MIDI track to the end of a selected track.

New Tempo Track

With this command you can insert a Tempo Track (view page 403).

New Folder Track

This command adds a folder track in front of the selected track. If a range across several tracks is selected before creating the folder track, each of these tracks will be added to the new folder track.

> You can find detailed information about folder tracks in the section "Working in the Project Window" > "Folder Tracks" (view page 136).

New Submix Bus

Add a submix bus as a track here. The new bus will be added below the selected track.

New AUX Bus

Add an AUX bus (view page Fehler! Textmarke nicht definiert.) as a track here. The new bus will be added below the selected track.

New Surround Bus

A surround bus corresponds to a normal submix bus (view page 225) with surround functionality. All tracks routed to a surround bus receive the Surround Editor instead of the normal Panorama knob, which can be used to adjust the surround position of the output signal of this track.

The Surround master is created simultaneously when a Surround bus is created; the individual channels are routed to the different output devices.If no Surround Master is present in the project when a Surround Bus is created, a Surround Master is created at the same time, whose individual channels are routed to the various hardware outputs. The Surround Settings dialog (view page 298) is called automatically for this purpose.

New Surround AUX Bus

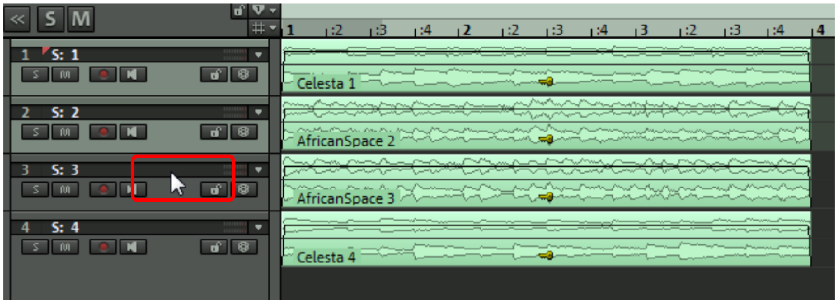
Use this command to create a new Surround AUX Bus.

A Surround AUX bus allows integration of Surround effects. This may be fed by a normal track or by a lower lying Surround bus. The AUX bus itself can again feed a Surround bus with a higher track number.

Copy Track(s)

Use this function to copy selected tracks to the clipboard to paste them into the same or another project (see below).

Individual tracks are selected by clicking on the track header (view page 139). To select more than one track, hold down the **Ctrl key** and click on the **track names**. To select multiple consecutive tracks, select the first track and hold down the Shift key to click the track name of the last track.



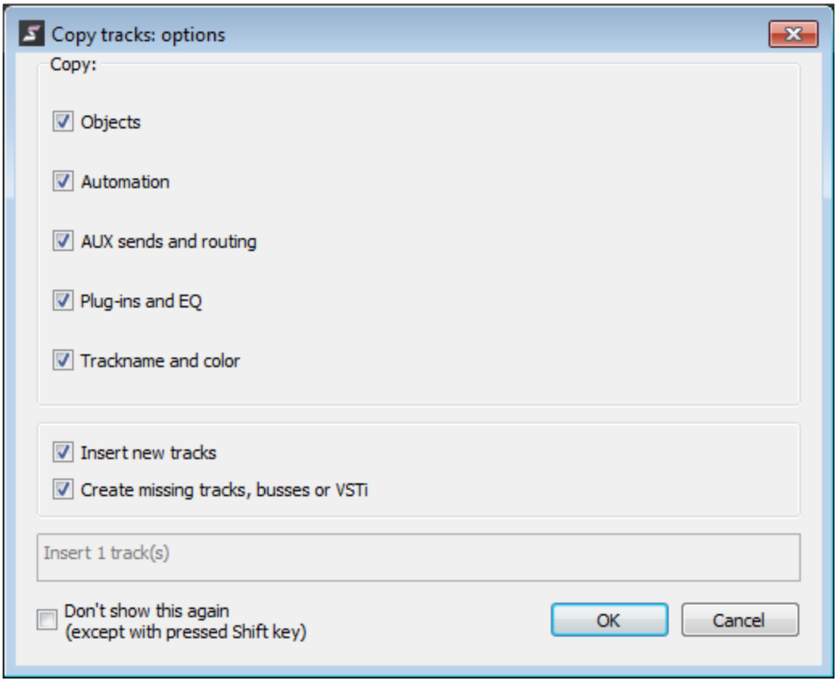
All track properties and contents are copied. When pasting (see the following section), you can specify which of these are to be inserted.

Paste Track(s)

This function inserts all tracks copied into the clipboard under the selected track.

At first a dialog with several options is opened. By this options you can use this function also to transfer properties of one track like plug-ins, automation or routings to another track, keeping the objects on this track untouched.

The dialog remembers the settings, therefore, for further insert operations with the selected settings you can skip the opening of the Options dialog box by setting the check mark **Don't show this again** in the lower right corner of the dialog box. To reach the options dialog again, hold the **Shift key** while chosing the Paste Track(s) menu item.



Copy: Here you can specify which track properties and contents are to be inserted.

- Objects: Any objects including their effects and automations
- Automation: all automation curves of the track. Note that all automation curves are inserted. If you have automated plug-ins in the copied track but do not copy the plug-ins, you get automation curves without a target. These are displayed in the automation menu with ???.
- Aux sends and routing: The values of the aux send faders as well as the audio and MIDI input and output routing. Note also the "Insert missing tracks, busses and VSTi" options below.
- Plugin-ins and EQ: All track effects as well as the plug-in instruments and effects loaded into the track.
- Track names and colors: The track names are important in that the assignment of the copied routing is based on the track names.

Insert new tracks: If you uncheck this box, no new tracks will be inserted, but the target project will be overwritten with the clipboard content. If exactly one track is selected in the target project, the target project will be overwritten with all the tracks from the clipboard. However, if multiple tracks are selected in the target project, only these selected tracks will be overwritten.

Add missing tracks, buses and VSTi: If you paste the copied tracks into another project and the "Aux sends and routing" option is active, this option ensures that the routing targets, aux and sub-mix busses are created if they are not copied with. All track settings of the additional tracks (e. g. plug-ins in the AUX busses, VSTi plug-ins as MIDI Out targets of tracks with MIDI objects, audio tracks for single outputs of VSTis) are taken over.

Attention: AUX and SubMix busses are only created if there aren't already busses with the same name in the target project. Otherwise these will be used, but the effect settings of the busses from the original project will not be taken over!

Note: The audio tracks of VSTi single outputs are also created including their insert and aux send effects, with additional AUX busses with effects if necessary. However, if these tracks are again routed to busses, these busses are no longer copied.

Delete Track(s)

This function deletes all selected tracks.

Track type

Here you can specify whether the track should be an **audio** track or a **MIDI track**. Moreover the track can be defined as a **submix** and/or **AUX bus**.

So you can define the track as **Economy Track.** (view page 79)

Input

Stereo in: The channel input is switched to stereo and the signal will be recorded on two channels.

Mono in: The channel input is switched to mono and the signal will be recorded on only one channel.

Mono In / Stereo FX: The track input is switched to mono and a single-channel input signal is recorded, but the track itself remains a stereo track in which the effect calculation is performed in stereo.

MIDI track record: If you click this menu entry the MIDI section of the track editor will open, allowing you to record MIDI files.

VST MIDI out recording: By activating the option "VST MIDI Out recording" on a track you can record all received VSTi MIDI Out data in the corresponding track.

Group track controls

With this function you can group control functions like faders, record or mute buttons from different tracks. If you want to group track controls, select the desired elements while holding down the "Ctrl" key and then activate "Group track controls". Control groups will be automatically called up when one of the elements of this group is activated.

Example: When muting a submix bus, no AUX channels from the original tracks will be sent. An additional control group can be created in order to simultaneously control the faders of several tracks. By clicking and simultaneously using the Shift + Ctrl keyboard shortcut individual faders can be moved in the opposite direction. By clicking together with the shortcut "Shift + Ctrl", individual faders may even be moved in opposite directions.

Note: Please note that only one control element may be present per control group. Overlapping control groups can't exist.

Ungroup track control group

You can use this function to ungroup. To do so, click on an element in the control element and select "Ungroup track control".

Hide Track

This command prevents the selected track from being displayed in the track view. However, it will still remain visible in the mixer and will continue to play there.

The menu item "Unhide All Tracks" shows this track and all other hidden tracks.

An overview of all tracks including their status of visibility is included in the Track Manager. ("View" > "Manager" > "Track Manager") Hidden tracks can be recognized in the track manager by the lack of a tick beside "Arrangement". In the column beside it to the right, you can hide the track in the mixer window as well.

Detailed information about the track manager is provided in the "Managers -> Track manager" section.

Unhide All Tracks

Use this command to show all hidden tracks in the arranger once again.

Track size

Maximize Track

The activated track is maximized, i.e. it is displayed in a large view.

Minimize None

All minimized (and maximized tracks) are displayed in the same size.

Automatically maximize track

The track selected by clicking on the track header is automatically maximized. This useful when there are a lot of tracks in the arrangement and you want to keep an overview of all peak meters, but still want to be able to access control elements for a track.

Display Subtracks

This command opens the subtracks for a folder track.

Track Freeze

This option renders the selected track as a wave file which replaces all the objects on the active track. The object and track effects are also included which reduces the CPU load resulting from CPU intensive effects and plug-in calculations.

The advantage of Track Freeze is that the frozen track is saved in a separate VIP and can be edited or recalled at any time. This allows you to work flexibly and reduce the load on your processor at the same time.

> Detailed information about freezing plug-ins is provided in "Software instruments/VST plug-ins -> Freezing plug-ins (Freeze)".

Track Freeze for AUX Buses/Submix Buses

Submix and Aux bus tracks may also be frozen. This way, you can get a complete group of tracks to release the required CPU load including all effects, fades, crossfades, and automation settings in one step.

Submix Bus

- Tracks routed onto the submix bus are not changed. They remain the same.
- The submix bus input is muted

- The file created while freezing is inserted into the submix track as an audio object.
- Changes made to the tracks after freezing which are routed to the submix bus have no effect on the playback of the frozen submix bus.

AUX Bus

- AUX send settings in tracks are taken into account when freezing
- Tracks routed onto the frozen AUX bus are not changed but remain unchanged instead
- The AUX bus input is muted
- The AUX send fader of the affected tracks is colored blue.
- An asterisk is added in front of the name of the AUX send fader in the mixer.
- The file created while freezing is inserted into the AUX track as an audio object
- After freezing, any changes made to the tracks that transmit to the AUX bus are not updated in the file that was created during the freezing process.

Edit Track Freeze

This command opens the temporary trackbouncing VIP in which the frozen track is saved.

You can make any changes you like here (just like in any other project), from object cuts via effect calculations in the track and object to adding other wave projects.

Frozen Submix/AUX Bus

If you have changed something on tracks that route to a frozen submix bus, you should "unfreeze" the bus using the "Track unfreeze" command and then refreeze it immediately afterwards with "Track freeze" in order to keep it up-to-date.

Unfreeze Track

This menu item removes the wave file created during "Track freeze" and brings back the track that was saved in the temporary trackbouncing VIP.

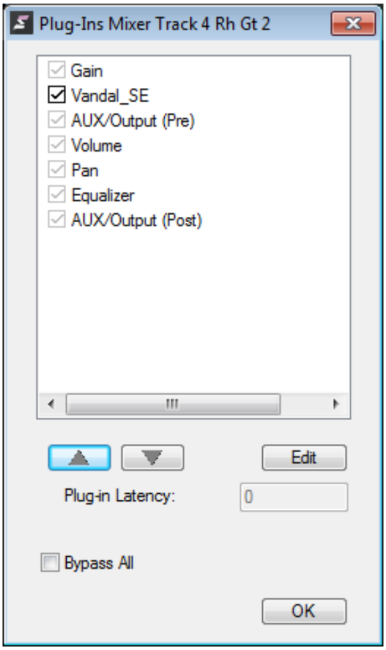
Changes made in the temporary freeze project are also applied to the track.

Note: If you add additional tracks which were created when freezing a track to the project, "Track unfreeze" will no longer be usable.

Track Effect Settings

Effect Routing Dialog

In the Effect Routing dialog you can make all important settings for realtime effects.



Effect Routing Dialog - Routing

All available effects are listed in the effects list in the respective context (track, object, or master). By right-clicking you can directly access the settings dialog of the selected effect. The check boxes for volume, pan, and gain knobs as well as the AUX gain knobs are grayed out because they are only represented in the list for the purpose of ordering the effects.

The order of the effects in the mixer tracks and the mixer master can be configured freely; in the object, it is restricted. You can change the position of an effect or plug-in with the up/down arrows.

On the tracks there are two AUX Send options (Pre and Post). A right-click opens the AUX routing dialog. Depending on whether the AUX send controller is switched to pre or post, the corresponding entry in the signal chain will be used to feed the AUX signal to the AUX bus. VST plug-ins can also be routed pre and post in a similarly flexible manner.

Effect Routing Dialog - Buttons

Edit: Opens the dialog of the active effect. This can also be done by right-clicking on the desired list entry.

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

Copy Track Effect Settings

This command allows you to copy the complete effects settings of the active track into the clipboard. This includes the order and parameter adjustments of all internal effects including those of VST plug-ins.

Insert Track Effect Settings

This command allows you to insert the complete effects settings from the clipboard in a track. This includes the order and parameter adjustments of all internal effects including those of VST plug-ins.

Reset Track Effect Settings

This command resets all track effects settings to default values, i.e. no effects are applied.

Save Track Effect Settings...

Here you can save custom track effect setting as .TRK files.

Load Track Effect Settings

Here you can load previously saved/preset track effect settings in the selected track.

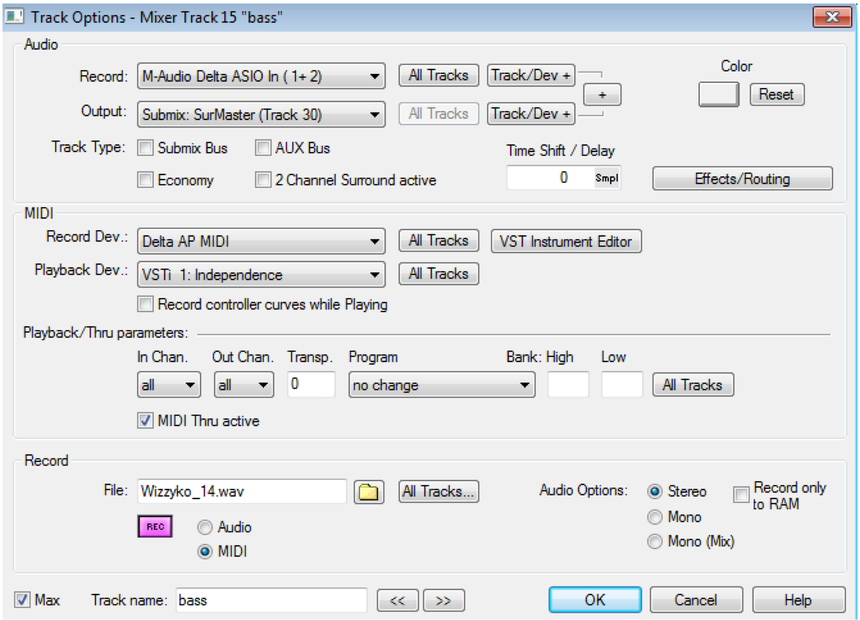
Mono Effect Processing

Effects for this track will be calculated in mono.

VST MIDI out recording

By activating the option "VST MIDI Out recording" on a track you can record all received VSTi MIDI Out data in the corresponding track.

Track Options...



Audio

Change settings for recording and playback of audio tracks here. These settings also include settings for audio created by VST instruments.

Record: Select the sound card device to record this track here.

All Tracks: Applies the same recording device selected for this track to all tracks.

Track/Dev+: Sets the recording device for this tracks to the next, e.g. if inputs 1 + 2 of the sound card are set, it will be switched to 3 + 4. The button with the "+" sign to the right sets the recording and playback devices to the next ones simultaneously.

Playback: Select your sound card device to record this track. You can also select a bus track.

All Tracks: Applies the same record device chosen for this track to all tracks.

Track/Dev+: Sets the playback device to the next, e.g. if set earlier to outputs 1 + 2 of your sound card, this will be switched to 3 + 4. The button with the "+" sign to the right sets the recording and playback devices to the next ones simultaneously.

Track Type: This indicates if the track is a aux/submix bus. One track may also be both. "Economy (view page 79)" as well as "2-channel Surround" modes may also be activated for this track.

Timeshift/Delay: Use this field to set an offset for an entire track. Positive values cause the playback from this track and the playback marker display to be delayed by the amount entered. Negative values cause playback to occur earlier. All other tracks and the playback marker will be delayed in relation to the entered value. To the right of the input field, you can select the unit for the time delay.

Color: Select a color for the corresponding track and its objects.

Effects/Routing: This button opens the effects routing dialog for the corresponding track.

Track Name: Use this field to change the name of the current track.

MIDI

The MIDI playback section provides access to the various MIDI options.

Recording Device (MIDI): Use this menu to choose a MIDI input device.

All Tracks: Sets the recording device selected for this track to all other tracks.

VST Instrument Editor: This button opens the graphical interface for the connected VST instruments.

Playback Device (MIDI): In this menu you can choose your MIDI playback device. Here you can also select VST instruments as playback devices.

All Tracks: Sets the playback device selected for this track to all other tracks.

Record controller curves during playback: This option enables controller curves to be recorded during playback.

The following are available as additional playback options: In-channel, out-channel, transpose, program change, bank high, bank low. You can also activate **MIDI-Thru** for the affected track. This activates the speaker icon for the track.

Track Name: Use this field to change the name of the current track.

Recording

The section "recording" contains settings for recording from audio and MIDI.

File: Indicate the name of the file to be recorded here.

Assign file name automatically

Click "All tracks" on the right of the file name box and then select the desired command from the popup that opens:

- File name_track number: Transfers the file name of the track to all other tracks and adds the corresponding track numbers to them, for example, funky_01.wav, funky_02.wav, etc.
- Project name_track number: Transfers the project name to all tracks and adds the corresponding track number to them, for example, demo_01.wav, demo_02.wav.
- Track name: Sets the track name of each track as file names for the audio recording, for example, Drums.wav, Bass.wav, etc.
- Track number_track name: Sets the track number and the track name of each track as file names for the audio recording, for example, 01_Drums.wav, 02_Bass.wav, etc.
- Project name_track name: Transfers the project name to all tracks and adds the corresponding track name to them, for example, demo_Drums.wav, demo_Bass.wav, etc.
- Get track name from first object name: This option gives the track name the same name as the first object in that track.
- Get track name from first object file: This option gives the track name the same name as the first file in that track.

REC: Specify whether or not you want to record MIDI or audio here.

Audio options

You may also choose between recording in **mono** or **stereo** on the selected track. A **mono mix** of both device channels is also possible.

The option "**Recording only in RAM**" allows recording to be done directly in the system RAM.

Revolver Tracks

New empty revolver track: This command creates a new revolver track.

Before track objects are rearranged, select the option "**New Revolver Track (copy)**". The track objects will be copied as a new Revolver Track and a star appears before the track name. The original objects may now be edited or repositioned to produce a different version of the track.

Delete Revolver Track: This command deletes a Revolver Track.

Previous Revolver Track: This command displays the previous Revolver Track

Next Revolver Track: This command displays the next Revolver Track

More information on Revolvertracks (view page 142)...

More

Cut tracks

This function cuts out all tracks that are selected in the track header (view page 139). The corresponding tracks are stored in the clipboard in the meantime.

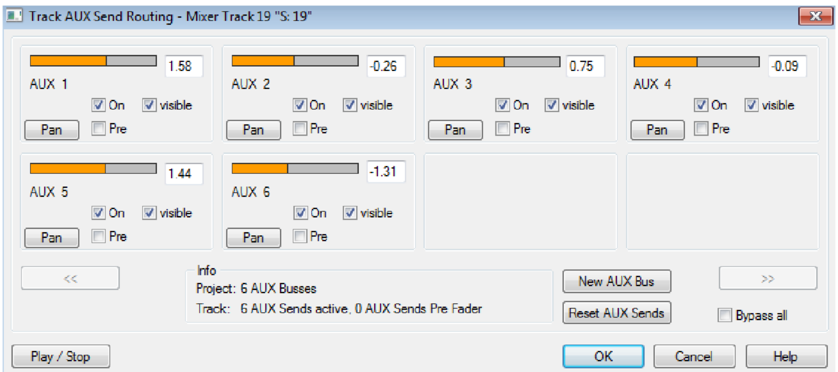
Track effects

These options allow (as with the Mixer channels) the application of real-time effects to the selected track in Samplitude.

More information about effects is available in "Effects - division and working methods (view page 236)".

AUX Sends

Use this dialog to create new AUX sends or determine the panorama out of existing AUX buses. You can also switch the AUX track to pre-fader or post-fader or deactivate it.



Description of the "AUX Sends" dialog

You can enter the send level of each AUX bus numerically or pull the orange bar in the gray field from left to right.

By default, all AUX buses are routed "post" in Samplitude. To switch them to "pre", activate the corresponding box.

New AUX Bus: Clicking this button creates a new AUX bus.

Reset AUX Sends: All AUX sends settings are reset.

Track Properties

This submenu provides access to all track properties.

Mute: Mutes the active track

Keyboard shortcut: Alt + M

Mute inactive: Mutes and also deactivates the track. This increases performance by removing the need for caching and FX processing for the track.

Keyboard shortcut: Ctrl + Alt + M

Solo: Switches the active track to Solo mode

Keyboard shortcut: Alt + S

Solo exclusive: This setting switches the active track to exclusive solo mode, i. e. only this active track is audible. All other tracks in solo mode are muted.

Keyboard shortcut: Shift + Alt + S

Solo safe: This setting ensures that all channels that are switched to solo are automatically monitored with the AUX Return channels that they feed.

Record: Arms track for recording.

Keyboard shortcut: Alt + R

Monitoring: Switches manual monitoring for the corresponding track on/off

Keyboard shortcut: Ctrl + Alt + Shift + F

Lock: Locks the active track.

Keyboard Shortcut: Alt + L

Edit volume: Here you can enter a numerical value in dB for the volume.

Shortcut: Ctrl + Shift + K

Edit panorama: Here you can enter a numerical value for the panorama curve.

Shortcut: Ctrl + Shift + P

Track phase inversion: Inverts the phase of the selected track.

Keyboard shortcut: H

Mute Bus Inputs: Sets the input of the selected Submix Bus or AUX bus to inactive, i. e. the signals routed to the bus are not routed through to the signal path of the bus.

Global Mute: The mute settings for all tracks are switched on or off.

Global solo: Enable/Disable all solo states of the tracks

Global: Solo-safe: This switches all tracks to solo-safe mode. In this mode, the corresponding track is monitored together with its AUX returns if switched to solo.

Global: Solo PFL: This switches all tracks to Solo PFL mode. In this mode, the corresponding track is accessed at a position before the fader and heard as a pre-fader signal if switched to solo.

Global: Solo exclusive: This switches all tracks to solo-exclusive mode. In this mode, the corresponding track is played exclusively if switched to solo. Tracks already switched to "Solo" will be removed from solo mode.

Pan/Surround Editor (Stereo-Panorama-Dialog)

This command opens the Surround/Panorama dialog which provides lots of useful panorama presets. If you are working in the normal Stereo Master Mode, you have the option of activating 2-channel surround.

The "Mono" button in the track editor sets the track to mono editing from input to the panorama controller. Particularly all track effects preceding the Pan controller operate in mono, which considerably saves CPU resources. The routing position of the Pan controller, however, can be freely adjusted in the FX routing dialog. In this case, the submix and AUX return buses always remain stereo.

If stereo objects are located in mono tracks, the mono share (L+R) is played.

If you use AUX sends, you can also use the panorama fader of the AUX send routing dialog for panning the mono signal.

In Samplitude there is also a Surround Master Mode which can be set up via the "File" menu -> "Project Properties" -> "Mixer Setup". If this mode is active, every track is set to Surround Mode.

VST Instrument Editor

Software instruments and MAGIX synths (e. g. Revolta) can be integrated seamlessly into a virtual project and controlled with internal MIDI functions and editors. The instruments are integrated into all effects and routing options. If you have a track open in a virtual instrument, you can open the user interface of the VSTi here.

Note: If activating this menu item does not produce a result, it means that a VST instrument has not been loaded into the corresponding track.

Detailed information about plug-ins can be found in the "Software/VST Instruments (view page 411)" chapter.

Activate Previous/Next Track

Use this command to activate the next (lower) or previous (upper) track in the VIP from the view of the currently activated track. With the cursor keys you can conveniently scroll up or down.

Shortcut:

Activate next track	Alt + cursor down
Activate previous track	Alt + cursor up

Track visualization

The visualization from the menu window can be set as a master or track display.

Detailed information on the individual visualization displays can be found in the chapter "View > Visualization" (view page 1011).

Object Menu

Unlike with the "Edit" menu all commands from this menu refer to selected objects in the virtual project.

Object Editor...

Use the object editor (view page 146) for object-oriented editing. This enables you to quickly and easily edit each selected object independent of any additional track settings.

Shortcut:	Ctrl + O

MIDI Editor...

This menu item opens the MIDI editor. The content of the MIDI object selected in the VIP may be displayed and edited here. You may choose from the matrix editor, drum editor, score editor, event list, and velocity/controller editor.

If a MIDI object is not selected, you will be asked if you wish to create one. If you confirm this with "OK", a MIDI object in the current track is created at the cursor position/start of the range.

Detailed information about this is provided in "MIDI in Samplitude -> MIDI editors (view page 328)".

Wave Editing

This opens the audio file associated with an object in the wave window. The range corresponds precisely to the part of the audio data that the object accesses.

More information about this can be found in the chapter "Samplitude Basics" > "Audio editing in Samplitude" (view page 66).

Destructive Editing

Destructive audio edit mode is the standard mode when the command "Edit a copy of wave content" in the Object menu is used to open audio files that belong to a VIP object.

Audio files (e. g. Wave files) opened directly through "File -> Open -> HD Wave" are opened in the virtual wave editor by default. This can be changed in the program settings (Y key, "Program" > "General > Open wave projects in destructive editing mode").

Regardless of the mode you load your wave project in, **you can switch between destructive and virtual wave editing through "File -> Project Properties -> Destructive Wave Edit Mode"**. If the box is checked, the extension "destructive" will appear in the title bar of your project.

In destructive mode audio files are edited directly on the hard drive. Changes made in the editing window are immediately applied to the audio file. To be able to undo actions in the opened file in this mode using "Ctrl + Z", you have to activate the undo function for audio files in the program settings ("Y -> Program -> Undo").

Note: At any time, a selected range from an audio file in destructive mode can be transferred to a VIP using "Copy" and "Paste".

Real-time Audio Editing

You can also load audio files without a VIP and edit them in a virtual real-time mode. To do this, deactivate "Open wave projects in destructive editing mode". in the program settings (Y key, "Program" > "General). Now the files that are loaded through "File > Import... > Load Audio file" can be opened in an audio file that works virtually similar to a VIP.

The advantage of this editing mode is that it combines real-time editing of the audio material with the features of non-destructive virtual editing that are the standard in VIPs.

In contrast to destructive editing you will notice the following differences:

The features "Cut", "Copy", "Delete", and "Insert" are non-destructive. The positions where these operations are performed are marked with dotted lines. These operations can be performed in real-time because it is no longer necessary to copy data to the hard drive for undo functions. The edits performed are only applied to the audio file when it is saved.

"Wave editing" mode – volume and panorama curve

Volume and panorama editing is also virtual in "Wave editing" mode. Activate "Volume Draw/Automation Draw" Mode. You can now draw level and panorama curves with the mouse, just like in the VIP.

Real-time Audio Editing - Master Section

If you open the mixer window by pressing the "M" key, you will see a minimized mixer on the master channel. All of the real-time effects that you are already familiar with from the VIP master section are now available for real-time audio editing.

This allows you to use several effects simultaneously during real-time editing. However, during destructive editing only one effect at a time can be written to the audio file.

"Wave editing" mode – auto-crossfade

The auto-crossfade option may be activated during any cut or insert operation in non-destructive wave projects

New MIDI Object

This creates a new MIDI object in the selected track. A pop-up menu opens when the function is selected to choose from multiple templates.

Presets include **MIDI drums** and **MIDI phrases** (filter sweep, 6 octave sequence, etc.) or an **empty template**.

These are also available in the "Templates" folder in the program directory. The pop-up menu may also be expanded with other templates simply by copying the desired MIDI files into the "Templates" folder or exporting them directly from Samplitude as a template.

New MIDI object in the range

This function creates a new MIDI object in the current track within the selected range.

New Synth Object

This command creates a 4-bar loop object at the current playback marker position. The instrument interface for the synth object that is created can be accessed by double-clicking on the object. Object synths include BeatBox 2, Loop Designer and Robota.

Edit

Cut Objects

Use this command to delete selected objects from the current project and insert them into the VirtClip. The content previously in the VirtClip will be overwritten.

Copy Objects

This command copies the selected objects into the VirtClip. The previous content in the VirtClip will be lost.

Insert Objects

Use this command to add objects from the VirtClip into the project, starting from the current position of the play cursorplayback marker this may cause objects to overlap.

Delete Objects

This deletes the selected objects from the current project. The content of the VirtClip will remain.

Shortcut: Ctrl + Del

Duplicate Objects

All of the selected objects are copied and positioned behind the original object dependent on the set grid.

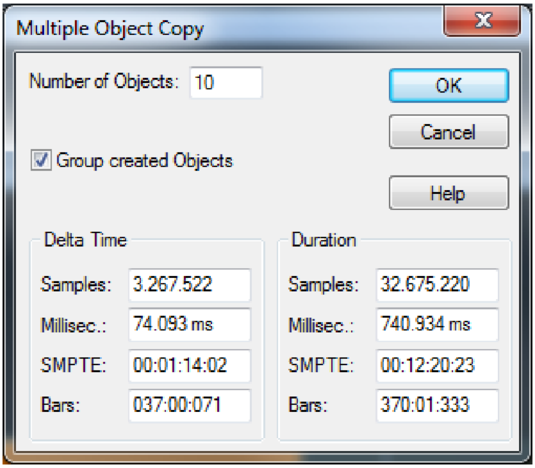
Duplicate Objects Multiple

Use this function to duplicate selected objects sequentially. In the dialog window that opens, enter the number of copies, the corresponding distance, and the entire length.

The following parameters are available:

Number of Objects: Specify the number of duplicates here.

Group Created Objects: Decide here whether the duplicated objects should be assigned to a group.



Time Difference: Set the distance between each object's starting point here. The default length is set to that of the object you wish to duplicate. This setting connects the objects seamlessly with one another so that a loop is created.

Duration: As an alternative to the time difference, the entire length of all copies may be set here. This is calculated from the product of the time difference and the number of copies.

Creating looped objects

Use this command to define a loop inside an object. Use the previously selected range above the object to set the loop start and end points. By dragging the length handle at the bottom right of the object you can make the loop exceed the previous object end point.

If you only selected an object, but not the range above it, the entire object will be looped by dragging to the right.

Loop objects are suited, for example, to quick creation of full drum tracks from a drum loop.

Loop points may also be set easily via the object editor.

Shortcut: Ctrl + L

Split Objects

Use this option to split all selected objects at the cursor position and separate them into individual objects.

If the range is selected, the cut will be made at each border. Ensure that the object that is to be edited is contained within the active range. If necessary, all objects may be activated by double clicking on the range selected in the range bar. In this case, it is not necessary to select the objects beforehand.

Shortcut: T

Split Objects at Project Markers

The object is split into separate objects at the marker positions within the object borders. The newly created objects will be named after the previous markers.

Split Objects at CD Track Indices

The object is split into separate objects at the marker positions within the object borders. The newly created objects will be named after the previous track markers.

Split Objects at Audio Markers

The object is split into separate objects at the audio marker positions within the object borders. The newly created objects will be named after the previous audio markers.

Heal/Unsplit Objects

This command recombines two objects that have been split without any audio calculation. To do this, the objects in the VIP must be positioned exactly beside each another, and the audio material must be connected. The objects must therefore be different but sequential pieces of audio material from the same wave project. If this not the case, the command will not function properly.

Before executing the command, select both of the objects that were split previously. All of the second object's properties, like fades and effects, will be discarded. The resulting combined object will feature the properties of the first object.

Trim Objects

This function moves the object borders of a selected object to the edges of the current range. To do this, the range must be contained completely within the object's borders.

Shortcut: Ctrl + T

Trim MIDI Objects

This function removes the MIDI data of an object that is not located within the current object's borders ("virtual MIDI data"). For example, this type of object virtual MIDI data results from dragging the left object border of a MIDI object after a MIDI recording inward (to the right). This function can be very useful if you wish to edit multiple MIDI objects at the same time in the MIDI editor, since overlapping virtual MIDI ranges can be very difficult to keep track of.

Glue Objects

A new object is created using those selected. Trackbouncing is used in this case. If you have selected just one object, this will be glued together with the object to the right. This function corresponds with the "Glue" symbol in toolbar.

Shortcut: Ctrl + Alt + G

After gluing, original cut may be subsequently edited the using the "Freeze object -> Edit object freeze" function. When selecting this function, a single-track VIP that contains the original objects will open as well. These may then be edited individually.

Use the "Freeze object -> Unfreeze object" function at any time to restore the original state of the object.

Mute Objects

This mutes the selected object.

Shortcut: Ctrl + M

Lock Objects

This protects objects from being unintentionally moved. When this function is active, the key symbol in the lower portion of the object turns red. If the key symbol is grayed out because of the display settings, the objects will be displayed crossed out instead.

Note: Please take note of the "Locking options". These specify which functions may be locked for set objects.

An object may also be locked by clicking the key symbol in the lower portion of the object.

Unlock Objects

This unlocks an object. All selected objects will be unlocked.

An object may also be unlocked by clicking the key symbol in the lower portion of the object.

Lock Definitions...

> Detailed information about this is available in "File -> Program Preferences -> Lock Definitions (view page 625)...".

Edit audio file copy

With this command you can create a copy of the audio file at the base of object, which you can then edit. You will then see the selected copy in the audio editing window to the right next to the original file.

Edit Root VIP

Audio objects that were compiled using an internal mixdown of a virtual project to a wave project and then inserted into a new project contain information about the VIP that was used to generate them (bounced). The "Edit root VIP" function allows you to reopen this VIP for editing.

Sample: A CD is compiled from any number of songs that were created in Samplitude and were converted to individual wave files using the trackbouncing function (view page 582) and are now visible in the arranger window of the new project. It's possible that you may want to change something else in the song retroactively. "Edit root VIP" opens the original project to make the desired changes. Once the modified project is saved, it will be re-bounced and updated in the project that is used to compile the CD.

Calculate MIDI Objects and Track Effects

This command permanently calculates all MIDI channel settings, transpositions and track effects into the MIDI data.

Demix MIDI Objects by MIDI Channels

This function will create a new track for each channel of the selected MIDI object. The objects will then contain only the MIDI files of the specific MIDI channel.

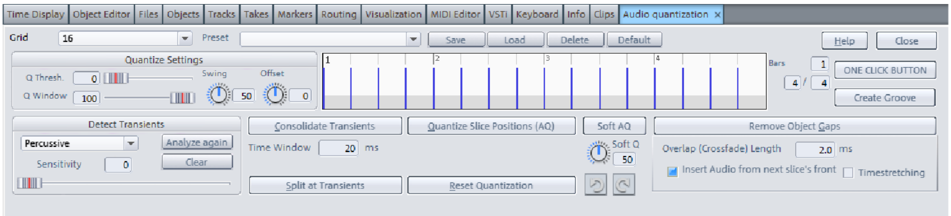
Extract Controller Curves from MIDI Objects

This option extracts the continuous controller data (MIDI CC) from MIDI objects and transfers it to the VIP as track automation.

Quantization

Audio quantization

Audio quantization is a special function for adjusting and snapping a recording to a grid.



These commands reflect the typical working procedure when "adapting" a multitrack drum recording to the VIP grid and are applied to the object selection. The "One click button" automatically executes the "Determine transients", "Consolidate transients", "Cut at transients", "Quantize object positions (AQ)", and "Remove Object Gaps" commands sequentially.

During determination of transients, you can set the sensitivity of the detection with the "Sensitivity" controller. The higher the value, the more beat markers are created.

By using "Recalculate" and "Delete", you can recalculate or delete an already created audio material analysis.

If you would like to connect the slices you have created to crossfades, then enter the length of the fade into the respective field "Overlap (crossfade) length".

Several gaps between slices can be closed either by object time stretching or applying audio material from the slice to the right of the gap.

> For a detailed description of the audio quantization wizard, see the menu reference > "Object" menu in the manual PDF.

Define transients

Audio quantizing objects is particularly well suited for modifying multi track drum recordings.

"F3" and "F2" allow you to jump to the next or previous transients in the audio signal. By quantizing the VIP object position and dividing objects at beat markers/transient positions you have the option to perform highly flexible dynamic drum quantizations.

All following commands refer to selected objects and can also be accessed using the Audio quantization wizard.

"Specify transients" marks the peaks of the selected objects as "AQ" with special beat markers. A beat marker is an audio object marker, which is written into the audio file. To make these markers visible, turn on "Objects -> Transients (AQ)" ("Shift + Tab"). All other audio markers may be displayed via "Objects -> Audio markers".

> Detailed information about audio markers can be found under "Menu > Play/Rec > Markers > Audio markers".

Consolidate Transients

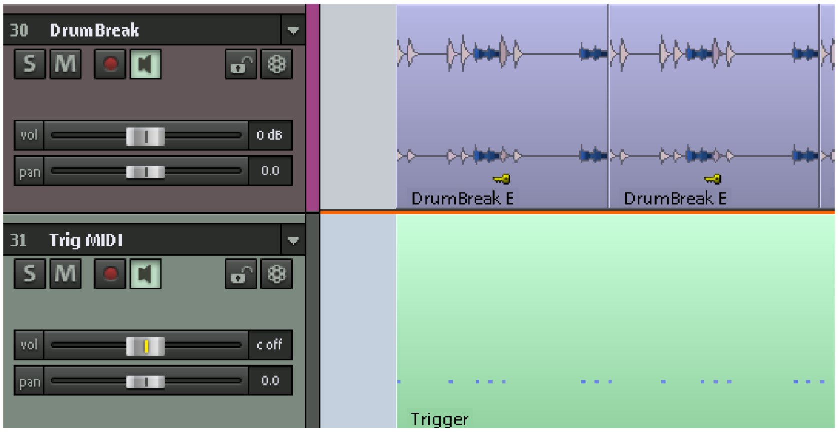
Transients within a certain range (e.g. 20 ms) are moved to the first transient position with this command. The effect is that "Split at transients" doesn't create too many splices, especially if you have several tracks with slightly different transient positions for the same drum hit, i.e. due to microphone distance related signal latency. The time window may be specified with the audio quantization wizard.

Create Groove

"Create groove" button in the audio quantization wizard creates a groove template based on the AQ markers (transients) according to the selected audio objects in the selected range.

Create MIDI trigger from transients

This function creates a new MIDI track below the track which contains the selected and analyzed objects. The transients found there will be displayed as MIDI events with corresponding velocity.



If you integrate a VST instrument in a newly created MIDI trigger track now you can double up or replace your existing audio track with additional sounds.

Create Groove Template from Transients

This function stores the transients created for the objects as a groove template in the program folder **"FX preset -> Grooves"**.

Select groove templates in the audio quantization wizard under "**Quantization settings -> Grid**". The length and start of groove template is always set to full beats.

Split at Transients

All selected objects are split over multiple tracks at the beat markers (AQ). This quantizes each drum hit individually.

Example: To split all drum tracks according to the bass drum and snare tracks, carry out transient recognition for the objects in those tracks. Next, select all drum tracks and apply the "Cut at transients" command. Now all objects will be split at any bass drum and snare hit.

Quantize object position (hard)

The menu command "**Hard quantize audio object position**" equals a soft Q value of "100" or "Quantize object positions (AQ) (view page 697)" in the Quantization wizard.

Soft Quantization of Object Position - Soft Q

This value sets the strength or "Soft Q" value of the quantization.

- "100" moves the event precisely to the quantize grid point,
- "50" shifts the event to the middle between the current position and the quantization grid point,
- "0" means no movement, i.e. quantization off

The menu command "**Hard quantize audio object position**" equals a soft Q value of "100" or "Quantize object positions (AQ)" in the Quantization Wizard.

Offset

The value range in this parameter stretches from -100 to +100. By changing the offset values, you move the whole quantization grid. Select a negative value for the offset, and the quantization grid will be moved by the corresponding value to the left, or forward in time. On the other hand, if you select a positive offset value, you will set the quantization grid by the corresponding value to the right, or backwards in time.

A value of -100 corresponds to a shift of half a grid length to the left; +100 corresponds to a shift of a half a grid length to the right.

Quantize object position (AQ)

Now all the selected objects will be adjusted to the corresponding quantization settings that you previously set in the Audio quantization wizard. During this process the **"Soft AQ parameter"** will not be observed.

When quantizing the object is hard quantized, i. e. it will be shifted to the next snap point.

If you are not satisfied with the results, you can "Reset Audio Object Quantization" settings, which will set the objects back to their original position

Soft AQ

"Soft AQ" executes soft quantization according to the "Soft Q" parameter settings.

Remove Object Gaps

After object quantization has been applied, gaps between the objects can occur, which may lead to audible dropouts in overhead signals. The command "Remove gaps between objects" provides options for closing these gaps. The function "**Use next object slice audio material**" may be used to move the object start position of the object to the right of the gap to the left until the gap is closed, or the gaps between the objects may be filled by **Time Stretching** the objects next to the gaps. Each method can be selected in the audio quantization wizard. The **overlap (crossfade) length** setting of the first option, i.e. "**Apply next audio object slice's audio material**", is applied.

Undo/Redo/Reset Quantization

Use the **round arrow buttons** to undo/redo the last quantization applied. In this case, the counter-clockwise round arrow serves as the "Undo quantization" function, while the clockwise arrow serves as the "Redo quantization" function.

Reset quantization: This function resets all quantization processes of the object positions.

Reset Quantization

The "Reset quantization" resets the selected objects to the original position and also undoes the quantization.

Quantize Settings - Preset

Different presets are available for selection:

- **5-tuplet**: Quantization occurs in fifths
- **Magnetic quantize**: The "window" value is set to "50", i.e. only 50% of quantization will be included. Only those events will be quantized that are located within a range of 25% of the snap range to the left and right of the grid point.
- **Soft quantize**: The "Level" parameter is set to "50", i.e. quantization occurs at a half interval between the current position and the next grid point.
- **Swing**: The swing parameter is set to "75", i.e. in contrast to the binary rhythm, which features a "swing" value of "50", uneven/un-highlighted counting times will be set to delay. This highlights the "swing" feeling.
- **Triplets**: Quantization occurs in triplets
- **16th offbeat**: The quantization grid's timing is moved back a 16th note
- **8th offbeat**: The quantization grid's timing is moved back a 8th note
- "**New groove**" and "**More life for hi-hat**" provide groove templates

Of course, you can also set your own values and save them as a preset.

Quantize Settings - Default

The "**Standard settings**" button provides the option of resetting the preset values:

Q threshold: 0

Q window: 100

Swing: 50

Soft Q: 25

Offset: 0

Quantization Setting - Grid

Use the "Snap/Grid" parameter to set the step length and thereby the quantization values for audio quantization. A value of "1" provides a snap value of a whole note, "2" is a half note, "4" is a quarter note, etc. Here there is also a selection of dotted notes, triplets and special groove templates.

Detailed information about groove templates is available in "MIDI editors -> Quantize -> Quantize settings - Groove template" (view page 343)

Q Threshold

The parameter "Q threshold" may be used to slightly vary quantization by excluding notes from quantization that are very close to the next quantization value.

Q Window

"Q window" refers to the interval to the left and right of a grid point; events will be quantized within this range. No quantization will take place beyond this, and for this reason, the events outside of this window will remain at their position. The quantization area is dependent on the values of the parameters "Q snap" and "Q threshold".

Example: Grid: 4 max. Window: 4

- 100: The Q range covers the entire area between the grid points on the quantization grid. All events will be quantized
- 50: The Q range covers half of the quantization interval. Events with gaps of ¼ of the grid width left and right of the grid point (1/16 note values in this example) are quantized
- 0: no Q range -> quantization off

Swing

Set swinging, ternary playing with this value. This enables you to enter the division for an uneven/unaccented grid points.

- 50: "50-50/1:1" split. The unaccented eighths are exactly half way between the even eighth notes ("even", binary playing method)
- 67: "67-33/2:1" tuplets. The beat is split up into three counts, whereby the note is assigned 2 beats (67%) and the off beat note one count (33%)
- 75: "75-25/3:1" split. For example, a pointed eighth and a sixteenth is created from two eighth notes

Show Grid/Time Signature

Displays the desired beat signature and number of beats. The display window will change accordingly.

Example of use:

1. Enable display for audio markers/transients in the view options (System options ("Y" key) > "Design" -> "View Options" -> "Objects" -> "Audio markers/Transients").
2. Load a VIP with a multitrack drum recording where the drummer has played to the click, so overall the VIP grid already matches the drummer's tempo.
3. Select a range for audio quantization. We recommend about 8 to 16 bars to guarantee a quick calculation. The analysis of transients for an entire song can take a long time. Trim the objects to the selected range size.
4. Select all drum objects, open the audio quantization dialog, and move the "Sensitivity" slider. Beat markers will now appear in the objects at the positions of the transients. Set the slider so that the main hits correspond to the beat markers. You can use the F2 and F3 keys to move the playback marker to the previous or next marker.
5. To cut objects for all tracks at the beginning of transients, specify the beat markers with the "Consolidate transients" command. Samplitude adjusts the beat markers of all tracks within a period defined in the "Time window" parameter to the beat marker position farthest to the left.
6. To separate all quantization objects, use the "Split at transients" command.
7. Use "Quantize object positions" to adjust the individual positions of the slice objects. You also have the option of changing the quantization settings here.
8. In some cases the quantization will result in gaps between the slice objects. These can be filled with the option "Remove Object Gaps". The relevant objects will be stretched.

MIDI Quantization (Standard)

This command conducts standard quantization of the lengths of the MIDI notes of all selected MIDI objects according to the MIDI quantize settings.

> Detailed information about quantization can be found in the chapter "MIDI in Samplitude > Quantization".

MIDI Humanize Q

This command applies the current **Humanize** value in the quantization options.

> Detailed information about quantization can be found in the chapter "MIDI in Samplitude > Quantization".

MIDI Quantization Settings

This command opens the dialog for MIDI event quantization.

Detailed information about quantization can be found in the chapter "MIDI in Samplitude > Quantization".

Extended MIDI Quantize

Start Q / MIDI Start Quantization

This command quantizes the start positions of the MIDI notes for all selected MIDI objects according to the MIDI quantization settings. The grid view corresponds to the start quantization value.

Detailed information about quantization can be found in the chapter "MIDI in Samplitude > Quantization".

Start and Length Q / MIDI Quantization (Start and Length)

This command quantizes the start and length of the MIDI notes for all selected MIDI objects according to the MIDI quantization settings.

Detailed information about quantization can be found in the chapter "MIDI in Samplitude > Quantization".

Length Q / MIDI Length Quantization

This command quantizes the length of the MIDI notes for all selected MIDI objects according to the MIDI quantization settings.

Detailed information about quantization can be found in the chapter "MIDI in Samplitude > Quantization".

Soft Q (approximation quantization)

This command lets you can take the current Soft Q (power) value into consideration while quantizing.

Detailed information about quantization can be found in the chapter "MIDI in Samplitude > Quantization".

Quantize Note Ends

This command quantizes the end of the MIDI notes for all selected MIDI objects according to the MIDI quantization settings.

Detailed information about quantization can be found in the chapter "MIDI in Samplitude > Quantization".

Reset MIDI Quantization

Use this command to restore the original offset of start and length values of MIDI notes for all selected MIDI objects. This enables you to undo quantization at any time (even after the VIP has been saved).

Detailed information about quantization can be found in the chapter "MIDI in Samplitude > Quantization".

MIDI Input Q (global)

Use this option to quantize the MIDI recording destructively. The original position may be restored via the menu item "Object -> Quantize -> Reset Quantization".

Select objects

Select objects under playback marker/range

Selects all objects found under the playback marker or selected track ranges.

Select all objects

Selects all objects in the arranger window.

Shortcut: Ctrl + A

Select objects in active track

All objects in the active track will be selected. This function may also be applied by double clicking a free area in the active track.

Object lasso

To select several objects, click on an empty area in the lower portion of the track and then drag the mouse over the objects you wish to select. Sometimes there is not enough space between the objects. In this case, adjust the mouse behavior with the object lasso.

Once activated, click on an object and pull up a selection frame without moving the object you clicked on (which would be the case in "Universal" mode). After selecting, the mouse reverts to regular behavior. To use the lasso selection method several times in sequence, the object lasso must be reactivated each time.

Shortcut: Ctrl + Alt + L

Select previous object

Use this command to select the previous object on the same track.

Shortcut: Ctrl + Alt + Q, <

Select next object

Use this command to select the next object on the same track.

Shortcut: Ctrl + Alt + W, >

Select to previous object

Use this command to add the previous object on the same track to the current selection.

Shortcut: Ctrl + Alt + Shift + Q

Select to next object

Use this command to add the next object on the same track to the current selection.

Shortcut: Ctrl + Alt + Shift + W

Switch selection

All deselected objects are selected and all selected objects are deselected.

Deselect objects

All selected objects are deselected.

Shortcut: Ctrl + Shift + A

Groups

Group Objects

This function assigns all selected objects to a group. As soon as one object is selected, all objects in the group are automatically selected so that you can work on them collectively.

Shortcut: Ctrl + G

Ungroup Objects

Use this function to ungroup a selection of objects. This turns all selected objects into independent objects again.

Shortcut: Ctrl + U

Temporarily Exclude Object from Group

This command removes most recently clicked object from an existing group. By clicking an object again and repeating the function, the removed object is moved back into the group.

Shortcut: Ctrl + Shift + U

or Shift + Ungroup button

Temporarily Exclude All Objects from Groups

This command temporarily removes all objects from their groups. In this case, the "Ungroup" button will blink. If the function is accessed again or if the blinking buttons are pressed again, the groups will be restored and the button will stop blinking and return to inactive status.

Shortcut: Shift + Alt + Ungroup button

The shortcut "Ctrl + Alt + Ungroup" resets the grouping history.

Additional information about grouping objects is available in the chapter "Working with objects -> Group/ungroup objects (view page 166)"

Moving objects

Move objects

This opens the dialog window to enter the starting position of an object in number of samples, milliseconds, SMPTE, or beat units.

Arrange Objects

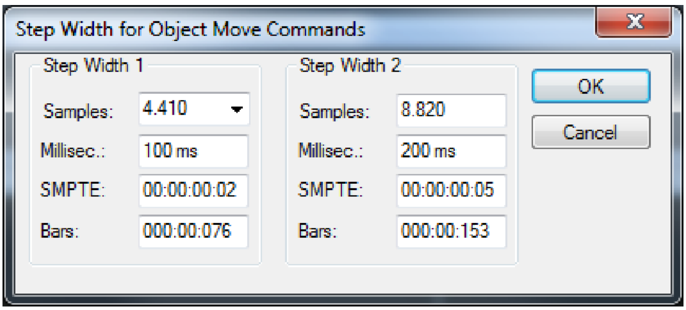
Use this function to edit the numerical interval between the selected object and the object in front of it on the same track. With multi object selection the entire distance between all objects on a track can be set.

Object/Fade Step Size

Select an object for simple object editing.

If you have selected two objects for editing, it is possible to edit it them separately as well as together.

Each of the commands below is available in two step sizes. The step sizes can be set in the dialog that follows.



Use the **"Ctrl" key to select the left object** for editing and the **"Alt" key to select the right object**. During this process the commands below are automatically carried out with **Step size 1**. To edit with a **step size of two**, **hold down the Shift key** as well. Now press a key on the number pad (0-9) to activate the respective function:

Left object to the left	Ctrl + 1
Right object to the left	Alt + 1
Left object to the right	Ctrl + 2
Right object to the right	Alt + 2
Object(s) to the left	Ctrl + Alt + +1

Object(s) to the right	Ctrl + Alt + +2
Object(s) track downwards	Ctrl + Alt + Shift + Page down
Object(s) track upwards	Ctrl + Alt + Shift + Page up
Object start to the left	Ctrl + 3
Object start to the right	Ctrl + 4
Object end to the left	Alt + 3
Object end to the right	Alt + 4
Crossfade to the left	Ctrl + Alt + 3
Crossfade to the right	Ctrl + Alt + 4
Fade-in handles to the left	Ctrl + 5
Fade-in handles to the right	Ctrl + 6
Fade-out handles to the left	Alt + 5
Fade-out handles to the right	Alt + 6
Increase left volume	Ctrl + 8
Decrease left volume	Ctrl + 7
Increase right volume	Alt + 8
Decrease right volume	Alt + 7
Increase volume	Ctrl + Alt + 8
Decrease volume	Ctrl + Alt + 7
Left object content to the left	Ctrl + 9
Left object content to the right	Ctrl + 0
Right object content to the left	Alt + 9
Right object content to the right	Alt + 0
Object(s) content to the left	Ctrl + Alt + 9
Object(s) content to the right	Ctrl + Alt + 0
Additional key for step size 2	Shift key

Please note the useful "**Object content to the left/right**" functions. Here the object length and object position are maintained while the assigned audio material from the corresponding audio file is inserted into it.

Object Hotspot to Play Cursor Position

Use this command to move the hotspot of the selected object to the playback marker position. Once you have activated a range, you can use this function to set the hotspot of the selected object to the start of the range. If you selected multiple objects, all other objects are sorted relative to the new position of the first object.

Shortcut: Ctrl + Alt + P

Object Begin to Play Cursor Position

Use this command to move the starting point of the selected object to the playback marker position. Once you have activated a range, you can use this function to set the starting point of the selected object to the start of the range. If you selected multiple objects, all other objects are sorted relative to the new position of the first object.

Object End to Play Cursor Position

Use this command to move the end of the selected object to the playback marker position. Once you have activated a range, you can use this function to set the end of the selected object to the start of the range. If you selected multiple objects, all other objects are sorted relative to the new position of the first object.

Object to Original Time Position

With this command you can move the selected objects to their original track positions.

Shortcut: Ctrl + Alt + O

Set New Original Time Position

This command sets the current object position as the new original position.

Edit Original Position

This command opens a dialog where you can edit the original position. To do this simply enter the new original position for the object in the input field.

Object step size 1

See "Object/Fade step size".

Object step size 2

See "Object/Fade step size".

Hotspot

Setting a Hotspot

With this function makes it possible to give an object another special snap point hotspot instead of its object front edge.

The hotspot is not placed at the current playback marker position inside a selected object.

Shortcut: Shift + H

A vertical line represents this in the object. From now on this object snaps to its hotspot at the corresponding snap markings.

You can make the selected object jump from its hotspot position to any changed position of the playback marker by using the function "Object hotspot to playback marker position (Shortcut Ctrl + Alt+ P)" (view page 707).

Delete Hotspot

This command deletes the hotspots for the selected object.

Takes

Take Manager...

Detailed information about the take manager is available in the chapter "Managers > Take manager (view page 189)".

Shortcut: Ctrl + Alt + Shift + T

Take Composer

This menu item opens the take composer.

Detailed information about the take composer can be found in the chapter "Manager -> Take Composer (view page 192)".

Object Color/Name

Object Name...

This command opens a dialog to assign names to all of the selected objects.

Shortcut: Ctrl + N

Object Background Color...

The selected background color from the color dialog is applied to all selected objects.

Object Foreground Color...

The selected foreground color from the color dialog is applied to all selected objects.

Note: If you have selected "Red/Blue alternation" or "WaveColor colors" for the waveform color in the display options ("View -> VIP display mode -> Definition (view page 632)"), there will be no change to the display of the object foreground color. The change to the waveform color will only be active when you restore the "Default color settings".

Spectral display

- The Spectral view is split for the left and right stereo channels (stereo representation)
- Activate Spectral view by going to "File > Program Preferences > System/Audio > Design > View options".
- Spectral display and WaveColor view are now available for individual VIP objects: "Object > Object color/name"
- Various color palettes are available
- Graphical data is saved as *hs files.

WaveColor Display

See „File“ > „Program Settings“ > „Display Options“ > „Waveform Color“.

Object Freeze

This function renders each selected object as a new wave file. The audio return signal of a software instrument replaces the original MIDI object with an audio object. In this case the audio return signal must be routed to the MIDI track.

The original object always remains preserved and can be re-edited using the "Edit Object Freeze" function or put back into the arranger with the command "Unfreeze Objects".

If more than one object is selected, the "Freeze Objects" function will be applied to each individual object. The frozen objects are grouped. Fade-in, fade-out, and object volume are not calculated, since these properties are applied to the object properties of every new resulting object.

Edit Object Freeze

This command opens the Freeze VIP where the frozen object is saved. This VIP contains tracks with the original object.

Changes made in the temporary freeze project are applied to the object as well if requested.

> **Note**: The length of the freeze VIP cannot be changed. This is because the length is set by the object to which the "object freeze" or "glue objects" function was applied.

Unfreeze Object

This menu item removes the wave file created during "Object freeze" and brings back the original object including the settings that were saved in the temporary trackbouncing VIP.

> **Note**: Do not create additional new tracks in the freeze VIP, since using the "Unfreeze object" function will no longer be possible.

Remix Agent – Tempo and Beat Recognition

The Remix Agent is a powerful tool for analyzing the tempo and beat of your music. First, automatic tempo and beat recognition is carried out for manual editing later. You can then split the object into remix objects, adjust the arrangement tempo and object tempo to one another, and write the tempo and beat information into the audio file.

Remix Agent – Applications

- Provides beat-precise splitting of songs so that you can rearrange the remix objects any way you like in the multitrack project.
- Adjusting the tempo of the arrangement to the tempo of the newly integrated song/CD track.

- Integration and adjustment of newly integrated song parts to the tempo of the arrangement

Remix Agent – Requirements

Tempo and beat recognition is applied on audio material with a length between 15 seconds and ten minutes. This should be rhythmic music.

Launch Remix Agent

Start the Remix Agent from the "Tools" menu or from the context menu by right clicking the object.

> **Note:** To use the Remix Agent in the wave editor, make sure that there is no tick beside "Destructive wave edit mode" ("File -> Project properties").

Remix Agent – Working Method

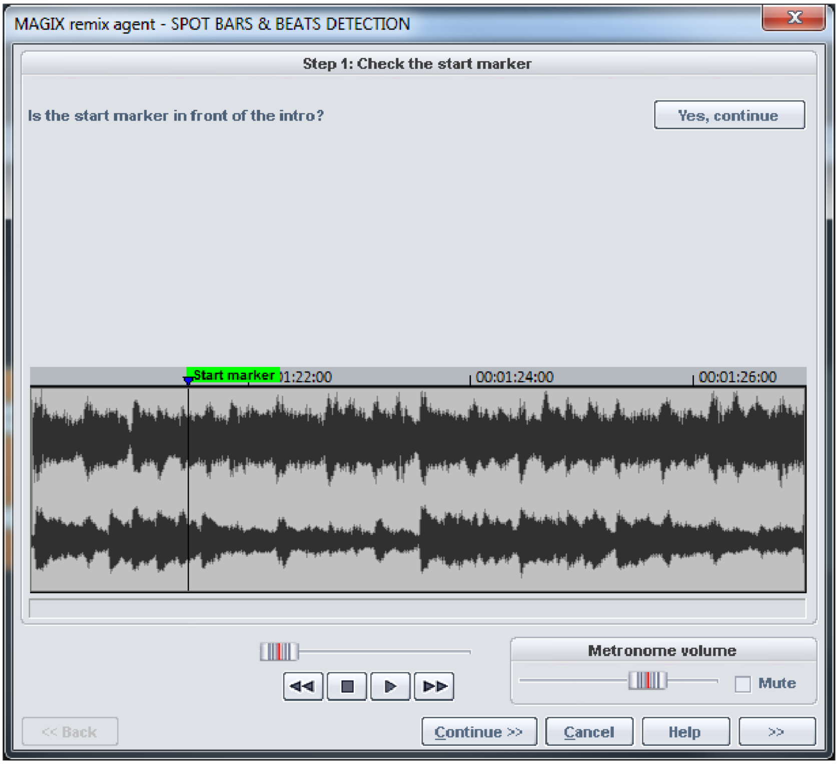
The Remix Agent works in fours steps:

Step 1: Checking the start marker

Step 2: Tempo recognition

Step 3: Specifying beat starts

Step 4: Application of BPM and beat recognition



Step 1: Checking the start marker

Place the playback marker at the position where detection should start in the remix object by clicking the timeline. If the object contains a longer intro without any beats or quiet synthesizer sounds in their place, the start marker should be placed after the intro.

The start marker should always be placed just before the beat or even better, just in front of the starting beat of a bar.

Alternatively, you can set the playback marker start position before opening the Remix Agent in the arranger of the virtual project or in the wave editor for the position where detection should start.

Step 2: Check the automatic tempo and bar recognition

After opening the Remix Agent by pressing the "Continue" button, it will begin to analyze the audio material and try to determine the tempo. The object is then played with the result sounding like a metronome click and numbered green beat lines appear in the waveform display.

Note: If the tempo or bar information of the object to be analyzed is already available, these are displayed as dots at the respective positions above the display of the waveform.

Below the waveform display, a display is provided on the left-hand side for the detected beat. In the middle, you'll find a small transport controller to simplify navigation. The fader acts as a position controller. To set the metronome volume, additional fader and mute buttons are provided on the right-hand side.

Correction of beat positions and tempo

Automatic tempo recognition doesn't always work from the start. If you don't hear the metronome clicking in time with the music, click the "No" button in the upper section of the dialog in order to access the manual tempo input dialog.

To correct the metronome speed and any time shifting that may occur between the metronome clicks, use the tempo correction as well as the "Tap tempo" button:

Tempo correction: The Remix Agent provides various speed settings; the speed the detected by Remix Agent as the most probable is preset. If the determined speed isn't correct, select a different, more suitable one from the list. The next time the object is played, it should be synchronous with the metronome clicking.

On/off beat correction: Sometimes the tempo is right, but the beats have been displaced. "On/off beat correction" provides a number of alternatives for moving the beats according to the complexity of the rhythm. Try out the different settings until you hear the metronome clicks in time with the beats.

Tap tempo: Alternatively, click rhythmically on the "Tap tempo" button or press the "T" key. Additional blue lines are displayed in the wave display. After at least four taps, the Remix Agent attempts to select the correct tempo from the list in "Tempo correction". The display next to the "Tap tempo" button displays the current status. Keep tapping until the red display showing "Unlocked" changes to the green "Locked" setting.

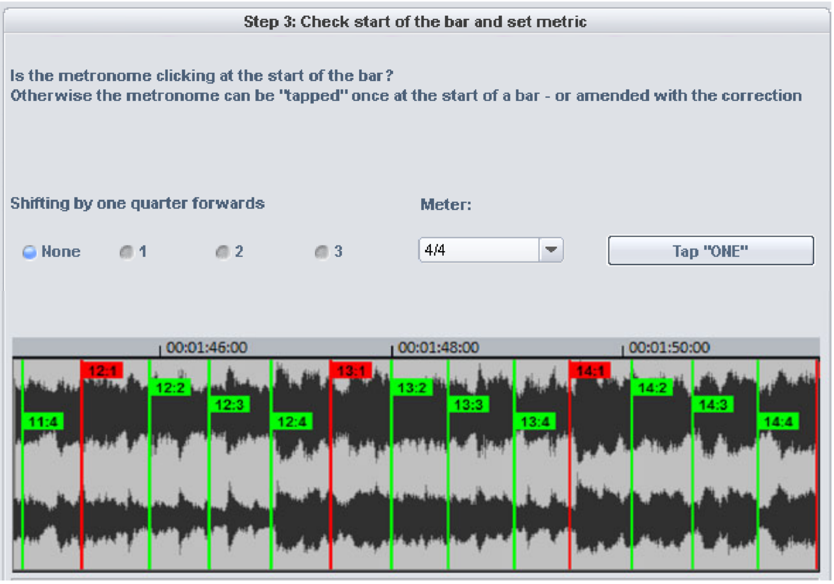
Use the **"0" key** to manually set the quarter beats while the music plays. Stray markers are automatically removed in so that the set tempo remains fundamentally intact.

You can move the markers with the mouse. If you hold the "Ctrl" button simultaneously, subsequent markers are also moved.

If the metronome clicks now correspond with the music, continue to the next step.

Step 3: Determining the start of a bar

Now set the beat type. 4/4 beat is set as default. You can correct it here if need be. The beat at the beginning of the beat should always be synchronized with the high, stressed metronome click or the red line in the waveform display.



This can be corrected in one step: If the start of the bar is audible, click "**Tap one**" once with the mouse or press the**"T" key** on the keyboard.

Alternatively, select how many quarter notes the "One" is to be moved back.

Use the **"0" key** to manually tap the position of the beginnings of the bars during playback. This is an efficient option for correcting the bars of longer sections.

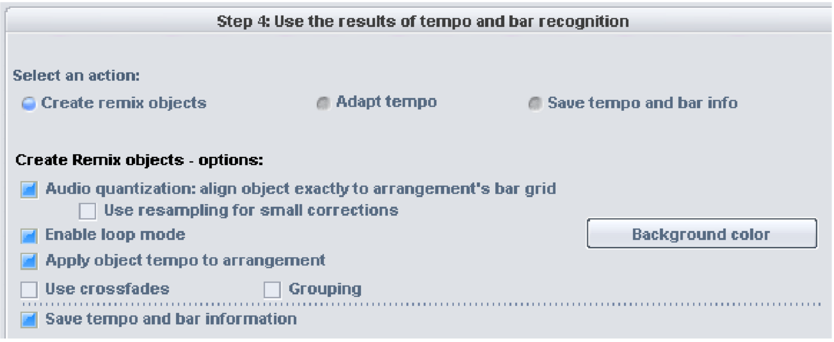
Continue to the last step once the starts of the bars are now correct.

Step 4: Application of BPM and beat recognition

This allows you to specify what should be done with the analyzed audio material. The following option is available:

- Create remix objects from the analyzed audio material
- Adapt the project tempo to the tempo of the audio material or vice versa
- Save only the tempo and beat information in the audio file for possible editing later on.

Create remix objects



Use this option to split the song into individual objects by beat, which can then be used later on in the virtual project.

Note: The "Create remix objects" option can only be processed if the Remix Agent has been opened from within the virtual project.

Create remix objects – audio quantization

If you have selected this option, the new objects will be fit directly into the beat grid of the arrangement.

Especially with live songs, there are slight tempo variations, making it possible that various beat lengths appear. Object Time Stretching is automatically activated and applied to correct the difference in length so that the objects still fit into the rigid beat grid of the arrangement.

The "**Use resampling for small corrections**" option makes sure that the higher quality "resampling" algorithm is used on smaller corrections rather than the "Time Stretching" algorithm.

Note: If you change the tempo of your multitrack project later and adjust the audio objects in the VIP to the new tempo, there will be clearly audible pitch changes in the remix objects.

Remix objects in "Loop" mode

If you select this option, the new objects will be set to "Loop" mode. This way, newly created remix objects can be stretched out as much as you like using the right object mouse handle.

Create remix objects – set arrangement tempo to the object tempo

Here, the arrangement of the virtual project takes on the found BPM value. To use the split song as the basis for a new composition in the remixes, use this option.

Use crossfades: Remix objects may be faded into one another with this function. The parameters of the transitions can be set in the crossfade editor.

Group: This function groups the remix objects.

Save tempo and beat information in audio file: If you select this option, the tempo and beat information are written to the audio file.

Background color: Click this button to set the background color of the remix objects you want to create.

Adapt tempo

This option provides you with the opportunity to adapt either the tempo of the analyzed audio material to your project or the project speed to the speed of the analyzed audio material.

Note: The "Tempo alignment" option can only be processed if the Remix Agent has been opened from within the virtual project.

Apply arrangement tempo to object tempo

This adapts the object length to that of the existing arrangement.

You can choose from three different methods:

Use Time Stretching: The pitch of the song remains constant in timestretching, but the sound quality may suffer.

Use Resampling: Resampling changes the pitch (similar to changing the speed of a record player), but retains most of the sound quality of the song.

> **Note**: If you change the tempo of your multitrack project later and adjust the audio objects in the VIP to the new tempo, there will be clearly audible pitch changes in the remix objects.

Use audio quantization: During audio quantization, the tempo adjustments are calculated into the audio file so that it seems as if remix objects were created and then immediately compiled into a new audio file. If recognition is unreliable, the result may feature extreme tempo variations. In this case, it is particularly important to set the start marker at such a position (before opening the Remix Agent) so that the tempo may be detected reliably. The advantage of audio quantization is that smaller tempo variations may be balanced in the music. The starts of the bars in the music correspond with the bar starts in the arrangement: they do not slowly drift apart.

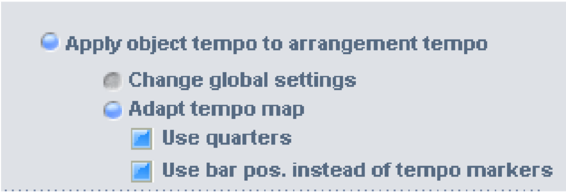
Save tempo & bar info into the audio file: If this option is active, the tempo and bar information are saved in the audio file. The objects in the virtual project won't be changed.

Set arrangement tempo to object tempo

Just like "Create remix objects", the arrangement of the virtual project assumes the BPM value determined in the Remix Agent. If you would like to use the split song as the basis for a new composition in the remixes, then select this option.

Change global settings: The BPM value of the arrangement in the VIP is set to the detected value.

Adapt tempo map: Sets a tempo marker in the virtual project's arrangement for each bar from the position of the playback marker to the end position of the remix objects.



- **Use quarter notes**: If this option is active, four markers are set (quarter notes) instead of one (whole notes).
- **Use beat position instead of tempo marker**: Beat position markers are set instead of tempo markers.

First example: Synchronizing a MIDI arrangement

1. Place the song object you want to synchronize the MIDI arrangement and the MIDI object to on top of one another on the same track in the virtual project. Open the remix agent for the song object. Specify the beat and quarter positions using the remix agent in step 1-3.
2. In step 4, select "**Adjust tempo**", then "**Apply arrangement to object tempo**" and "**Adjust tempo map**". This specifies that you want to create a tempo map.
3. Now select "**Use quarter beats**" and "**Bar position instead of tempo marker**". The tempo map receives a synchronization point at each quarter and not only at the beginning of each beat. If you click the "**Apply**" button and open the MIDI editor for the MIDI object, the beats will match one another and all notes will be shown and played in sync to the song. Even the metronome click will be in time with the song.

Second example: Mixing two song objects

1. First, adapt the project tempo of the virtual project to that of the first song object that you want to fade from. Select the options "**Apply object tempo to arrangement tempo**" and "**Change global settings**".
2. Next, adjust the speed of the second song object to that of the project by applying the "**Set object tempo to arrangement tempo**" option.
3. Since the project speed now matches the song speeds of both songs, you can easily fade between the two songs.

Save tempo and bar info

If this option is active, the tempo, and bar information are saved in the audio file. The objects in the virtual project won't be changed.

Generate beat markers in the current range: Use this option to place markers at the beginning of the beats of a song. This corresponds to the positions of the strokes displayed in red in the waveform display.

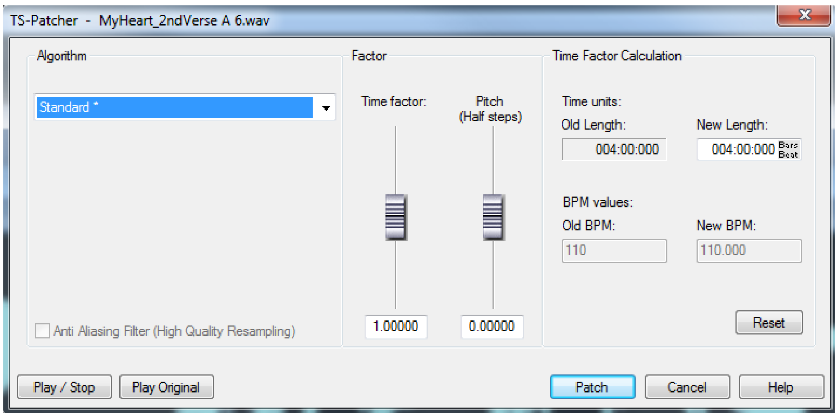
Generate quarter markers in the current range: Select this option to place markers at the beginning of the positions of the quarter beats starts. This corresponds to the positions of all the strokes in the waveform display.

WaveColor audio search

This function uses the WaveColor algorithm. This allows you to find the same or similar sounding ranges in an audio file very quickly. To do this, double click on the object you wish to use while holding down the "Shift" key, and this will then open it as a wave project. Now, select the range that you want to examine and copy it to the clipboard (keyboard shortcut: "C"). Now start the audio comparison function.

The dialog that opens contains a "Sensitivity" slider which lets you adjust the threshold value for the algorithm. Depending on the sensitivity level you choose, a different number of markers with the name "Match" will appear in your project at those points where the algorithm identifies similar-sounding passages. Close the search dialog. You can now jump to the ranges that were detected by clicking the corresponding "match" marker.

Time Stretch/Pitch Shift Patcher...



Time Stretch/Pitch Shift Patcher - Overview

Use this tool to patch wave files for use in Samplitude. In this case, additional information and settings may be written into the wave project. These enable the processing of Time Stretching and Pitch Shifting Algorithms at optimum quality.

Without the patcher, you would have to search and set the settings and additional information for each wave file you wish to edit.

The following may be patched individually:

- Algorithms for Time Stretching/Pitch Shifting
- Time factor/pitch
- Time factor processing (new length/new BPM)

Timestretch/Pitchshift Patcher - Patching Process

Patching only works when you have opened the audio file in audio editing (view page 66). It can also not be open in a virtual project at the same time.

Select the algorithm you want to use from the dialog. You can also specify the tempo of the wave file using the "Time Factor" controller.

You can now test the selected algorithm in conjunction with the "Play/stop" and "Play original" buttons. If you click the "Patch" button, Samplitude will write the information permanently to the wave file.

Algorithm for Time Stretching/Pitch Shifting

In Samplitude you can use the algorithms élastique Pro, élastique Efficient, Resample and Monophonic Voice for realtime time stretching and pitch shifting.

Detailed information about this is available in "Effects" > "Time / Pitch" > Resampling/Time Stretching (view page 808)". When you find a time stretching algorithm that meets your requirements you can use the Patcher to save it in the audio file. This ensures that this selected algorithm will be used automatically when you apply time stretching/pitch shifting to your audio files.

Beat Markers

Beat markers synchronize the audio material so that the groove remains perfectly intact. When using the beat marker-based algorithms, the beat markers are also saved in the wave project.

> **Note**: Unlike with the Time Stretching dialog in the "Effects" menu, the patcher is non-modal, meaning that you can move the beat markers in the wave project and simultaneously control the result of the Time Stretching.

BPM Value

The desired BPM value (beats per minute) can be patched here. This is useful for when the Time Stretching factor must be determined later to adjust the wave project to the tempo of an existing arrangement. If the "New BPM" field in the dialog is grayed out, you can change the value using the "Time factor" controller.

More

Duplicate Objects (in Place)

Use this function to duplicate all selected objects. The duplicated objects lie congruently over the original objects. The duplicated objects may be moved to the desired position by dragging the mouse in "Object/curve" or "Universal" mode.

Duplication is also possible via drag & drop while holding down "Ctrl".

This enables objects to be duplicated quickly without having to access the VirtClip.

Extract

Use this command to delete all unselected objects.

> **Note** the difference between this and the "Extract" function in the "Edit" menu (view page 652), which only applies to the selected area.

New Object

This function creates a new object. The last activated object is inserted at the position of the playback cursor in the selected track.

Play/Rec Menu

Play Once

This plays the wave project or the selected range once. This function corresponds to the "Play" button in the transport control (view page 90) as well as the "Play once" button in the toolbar.

Shortcut: Spacebar

Play Loop

This plays the wave project or the selected range in a loop. This function corresponds to the "Play" button in the transport control (view page 90) as well as the "Play loop" button in the toolbar.

Shortcut: Spacebar

Play in Range/Loop

The current range is played from the start of the project as a loop. This mode is especially useful for testing loops in the instrument sample and corresponds to the "Play in range/loop" button in the toolbar.

Shortcut: Shift + P

Play with Preload

Playback is prepared with all buffers loaded. Use this function prior to starting precise synchronization manually.

Shortcut: Shift + Spacebar

Play Only Selected Objects

Use this command to play the selected objects only. All unselected objects will be muted temporarily. If the play cursorplayback markeready located at the first selected object's time position, it will be moved there.

Shortcut: Ctrl + Spacebar

Play Cut

Play to Cut Start (In Point)

A defined time segment is played until the start of the selected range. The duration of this segment can be set under "Options -> Program Preferences -> Set Pre-roll time (view page 641)".

Keyboard shortcut: F5

Play from Cut Start (In Point)

A defined time segment is played from the start of the selected range. The duration of this segment can be set under "Options -> Program Preferences -> Set Pre-roll time (view page 641)".

Keyboard shortcut: F6

Play to cut end (Out Point)

A defined time segment is played until the start of the selected range. The duration of this segment can be set under "Options -> Program Preferences -> Set Pre-roll time (view page 641)".

Keyboard shortcut: F7

Play from Cut End (Out Point)

A defined time segment is played from the start of the selected range. The duration of this segment can be set under "Options -> Program Preferences -> Set Pre-roll time (view page 641)".

Keyboard shortcut: F8

Play Over Cut/Crossfade

Here a cut operation is simulated where a defined time range is played back up to the start of the selected range, the selected range is skipped over and a specified range after the end of the selected range is played back. The duration of the pre-roll can be set under "Options -> Program Preferences -> Set Pre-roll time (view page 641)".

Keyboard shortcut: F4

Play into Cut

Playback begins allowing for the lead-in time before the selected range starts and ends allowing for the lead-out time at the end of the selected range.

Restart Play

Use this command to move the playback marker to the original position during playback or back to the range start and then start playing from this position once again.

Stop

Choose this option to stop playback. The playback marker will jump to the start position. Whether or not the original position is the previous starting position or current stop position may be set in the playback parameters (shortcut: "P").

Shortcut: Spacebar

Stop and Go to Current Position

This option cancels playback; the playback marker will remain at the current position.

Shortcut: Number block "0" or ","

Playback Mode

Loop Mode

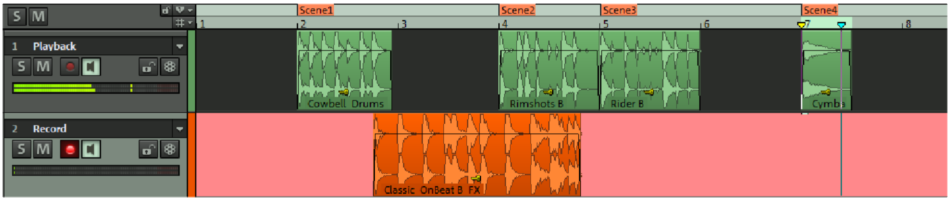
This mode plays the selected ranges in a loop. This corresponds to switching on the "Loop" button in the transport control.

Change Play Direction

This command changes the playback direction, even during playback.

Cue Mode

Cue mode is a special live playback mode often used for special cue sounds in radio broadcasts and theater. The objects (cues) in this audio track can be played back in sequence while another track is recorded simultaneously.



You can control cue mode in two ways:

1. Control using object edges

Starting playback with the space bar highlights the next range and then plays it in Cue mode. Playback stops automatically at the object's end. Upon resuming playback the following object starts. Related objects are treated as an object.

You can play back multitrack projects in cue mode. The object on the first track will be used for the Start/Stop range markings.

2. Control with track markers

With the help of the CD track markers you can use and control cue mode for complex arrangements independent of the object edges of the first track.

To do this, go to „System Options > Playback (view page **Fehler! Textmarke nicht definiert.**)“ and activate "At Track Indices" in the Cue Mode section.

Now position CD track markers at the desired positions in the timeline.

Once play is activated, playback will begin from the next CD track marker and end at the next one after that, which in turn is the starting put for the next playback range.

Note: For smooth cue mode functioning deactivate the loop mode.

In cue mode, the recording mode in the transport console changes to **Record without playback (band control)** (view page 730). The cue track is highlighted with a special background color.

If recording is done in cue mode, automatic scrolling in the arranger window will remain switched off.

The playback marker can be repositioned at any time within the currently played cues.

In "System Options -> Playback (view page Fehler! Textmarke nicht definiert.)" you can ensure that the **first track in cue mode is switched to "Solo"**. Here you can also set the **pre-roll (ms)** to determine the delay between the execution of the record command and the actual start of the recording.

Continuous playback while editing

The default behavior of Samplitude 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 during playback that require range selection, this behavior interferes 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.

Scrubbing Active

- Scrubbing Active Keyboard shortcut: "**Alt + Shift + Arrow down**"
- Jog (Absolute)
- Two Speed
- Shuttle (Relative)
- Scrub Left Keyboard shortcut: "**Alt + Shift + Arrow left**"
- Scrub Right Keyboard shortcut: "**Alt + Shift + Arrow right**"

Detailed information about scrubbing is provided in "Playback -> Playback options -> Scrubbing (view page 728)"

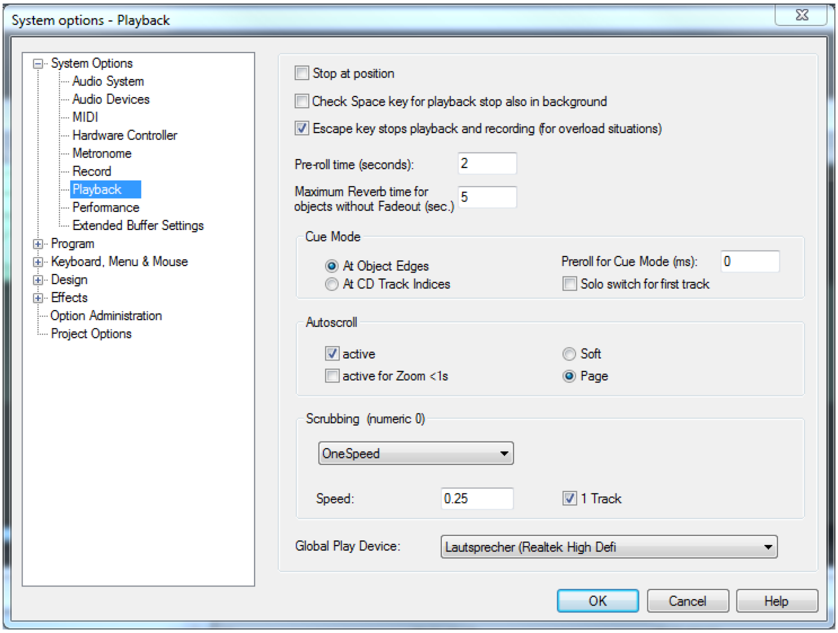
Playback at speed 1 - 4: Select from four different playback speeds. Playback begins immediately after the corresponding object is clicked.

- Play with Speed 1: Playback begins at the cursor position at 1/4 of the original speed.
- Play with Speed 2: Playback begins at the cursor position at half speed.
- Play with Speed 3: Playback begins at the cursor position at original speed.
- Play with Speed 4: Playback begins at the cursor position at double speed.

Recalling one of the four playback speeds with the Shift key pressed opens a dialog window, where you can specify an individual speed value.

Playback Parameters

This option opens the playback parameter window.



Keyboard shortcut: P

Check the "**Stop at current position**" option if you want the playback marker to remain at the same position after stopping playback. If this option is not active, the playback marker will jump to the original position or back to the start of the range when playback is stopped.

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

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

The **pre-roll time** is a necessary parameter for editing in Samplitude. Pre-roll time is used to define the time section in seconds which is to be played before the start of the selected range.

Here the **Maximum reverb time for objects with no fade-out (sec.)** can be set up to 60 seconds.

Note: Please note that long reverberation times can lead to performance problems.

To use **cue mode** you have to specify whether playback control will be carried out at the **object edges** or the **CD track indices** (track markers). You can also ensure that the **the first track in cue mode is set to „Solo"**. Here you can enter the **pre-roll in milliseconds** which determines the delay between the execution of the record command and the actual start of the recording.

Detailed information about Cue Mode is available in "Play/Rec > Playback mode > Cue Mode (view page 725)".

With **Autoscroll** the graphical display constantly moves even before the playback marker leaves the visible section. This provides a constant overview. Switch to Autoscroll Mode by checking "**Active**". "**Active for zoom < 1 s**" causes autoscrolling to begin at very high zoom levels of less than a second.

You can choose between "**Page**" and "**Soft"** autoscroll modes. With page scrolling, the section changes before the playback marker moves outside the section, while with soft scrolling, the playback marker always remains in the middle of your chosen section and the arrangement moves along beneath it. At smaller buffer sizes (view page 70) (e.g. < 4096 samples), scrolling will be softer.

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

Scrubbing

If you press "**0**" on the number pad or the key combination "**Alt + Shift + Page down**" and hold these keys down, Samplitude switches to "Scrub Mode".

This provides special control via the mouse over the selected track's playback speed.

The keyboard shortcut for scrubbing to the left is "**Alt + Shift + Left arrow**"

The keyboard shortcut for scrubbing to the right is "**Alt + Shift + Right arrow**"

There are four different scrubbing modes to choose from:

Shuttle: The relative distance between the playback marker and the mouse position is used to control the speed. The playback marker follows the movement of the mouse. The further you move the mouse from the playback marker, the faster playback will be. This means:

Scrub control faders at the left edge = double speed backwards,

Scrub control fader in the middle = no movement,

Scrub control faders at the right edge = double speed forwards.

Absolute: Use the absolute position of the mouse in the window to control the speed.

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

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

Scrubbing speed: This specifies the factor of the original speed applies to the scrubbing speed. The range of values stretches from 0.01 to 10.0, i.e. 1/100 of the original speed up to ten times the regular speed.

Scrubbing only on active track (1 track): If this check box is active, Samplitude will only scrub the active track.

Global play device: Specifies the sound card driver for playback.

Playback, Scrub and Varispeed Settings

Right-click the scrub control button in the transport console to open the „Playback, Scrub und Varispeed Settings“ (view page 621) dialog. In addition to the functions already mentioned, you can also adjust the sample rate, Varipitch and Varispeed settings here.

Record

If you select this menu point, recording is started immediately for the live, active track. You can record both the audio as well as the MIDI into each active track here.

Detailed information on recording can be found in the chapter "Samplitude Quickstart -> Workshop: recording (view page 40)".

Shortcut: R

Record Mode/Punch In

Standard Mode (Playback while Recording)

This option corresponds to the option in the recording options with the same name. This mode allows tracks to be recorded in addition to existing audio tracks while playback is running.

Switch on the new recording tracks and activate monitoring.

These tracks are now active in "Input" mode, i.e. the input sound desired for recording will be audible for these tracks. Recording begins immediately at the playback marker position when the recording button was clicked.

If you have selected a range behind the playback marker on the grid and marker bar and have activated "Loop" mode, then recording will only begin will only be activated once the start of the range is reached. Until this the track will signal that it is ready to record (the recording button will blink during playback), then the selected range will be recorded in "Loop" mode.

Record without Playback (Read After Write)

If you use "**Record without playback**" mode, recording will start at the playback marker position when the record button is clicked. In this case you will only here the input signal of the track that is in record mode and the playback marker will not follow recording in the timeline.

Now if you start playback from the position of the playback marker using the "Play" button on the transport console, you can hear all of the tracks being played back from this position. The recording continues independently of this until the "Record" button is clicked again. Now you will see the recording you just made as a new object in the arranger.

Punch Marker Mode

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

There are two principal approaches for punch recordings:

- **Punch "On-The-Fly"**: In this mode, the recording can be started (punch-in) and stopped again (punch-out) at any time during playback. Multiple punch processes can be made during a single pass in order to improve various positions in one recording. To do this, start playback with the spacebar. Now you can click the "Record" button in the transport console to "punch in" and "punch out" again.

Note: If you check "Prepare all tracks for track punch record" in the "System options -> Record (view page 606)" menu, you will also be able to punch your recording by using the recording buttons for the individual audio tracks (see below).

- **Punch with markers**: To execute the punch procedure with markers, activate the buttons "In" (sets the punch-in marker) and "Out" (sets the punch-out marker) at the desired positions. Start the process with the record button on the transport console or of the corresponding track. The actual recording will only take place within the punch range. When the recording is started and the playback marker is not yet in the punch range the "Record" button will blink. During punch recording, it remains red.
- **Multiple punch recordings in one pass**: It is possible to make multiple punch recordings in one pass with the help of punch markers. This is done by positioning the playback marker at the desired point and pressing the "In/Out" buttons in the transport control while pressing the "Alt" key.

Note: When working with punch-in/out, it is useful to have "Auto crossfade" mode active. "Auto crossfade" will create smooth transitions automatically between takes

- **Punch recordings** can also be executed **as loops**. Select a range across the planned punch section, press the "Loop" button and activate the recording. This will be looped until the process is canceled with the spacebar. Punch-in and punch-out markers are set automatically. A punch recording is at the punch position per pass. To find the best take from these looped runs, use the "Take manager (view page 189)".
- If the recording option "**Prepare all tracks for track-punch record**" is active ("System Options -> Record (view page 606)"), audio tracks that were not active when recording was started can be added to a recording, or tracks can be removed from the recording. To do this, click on the record button of the desired track. Punching on single track requires that the track has been assigned its own sound card input which is not already being used for recording. This is indicated by a red border around the record button in the track header.

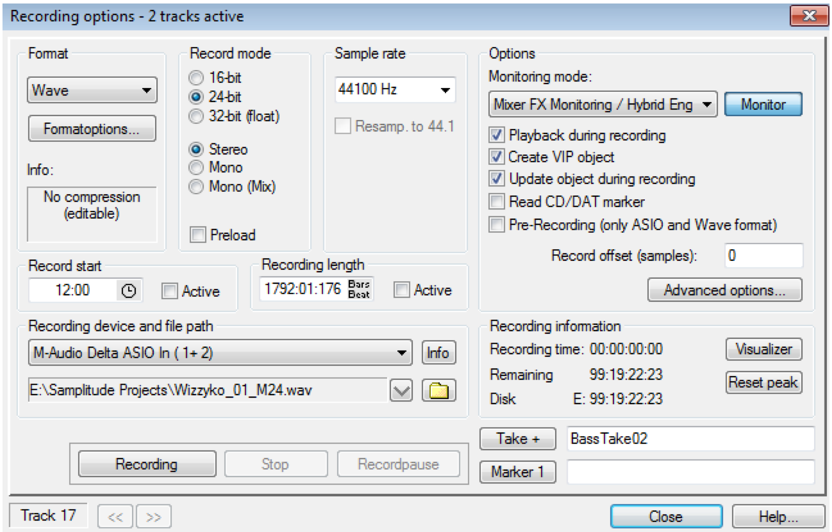
Note: Tracks that have been displaced in "Punch marker" mode by individual track punching will not be influenced by punch markers.

Record Pause

Recording is set to pause.

Recording Options

This menu option opens the Record window. All recording settings can be changed here.



Shortcut: Shift + R

Format: Set the corresponding recording file format (Wave, MP3, MPG, WMA, Real Audio, AIFF, Ogg Vorbis, and FLAC). In the infobox below, you'll find information about each of the formats available. This way you can see if the format is editable in Samplitude as well as the compression it uses.

Bit rate: Select the desired bit rate here for your recording.

Recording mode: Choose between stereo, mono, and mono mix.

- If you select "**Stereo**", the channel input will be switched to stereo, and the signal will be picked up on two channels.
- If you select "**Mono**", the channel input will be switched to mono, and the signal will be picked up only on one channel.
- If you select "**Mono (Mix)**", the channel input will be switched to mono, and the signal will be picked up on two channels, and then mixed together.

Preload: The recording is prepared and all buffers will be loaded. Thereafter, the message "Data loaded. Waiting to start..." appears. As soon as you press "OK", the recording starts without any delay.

Sample rate: This option allows you to select the sample rate of the recording. Take note that your audio card must support the selected sample rate.

Resampling to 44.1: If you have set a sample rate of 44,100 Hz and then activate this option, Samplitude will apply resampling at 44.1 kHz. The quality of resampling may be set via "**System options -> Effects -> Resampling/Bouncing.** (view page 615)".

Recording start: Set the recording start time in this field. The recording is then controlled automatically by the internal system clock to a specific time.

Recording length: Specify the recording length in this field. Using "Recording start" and "Recording length" you have the option of executing timed recordings even when away. "Recording" is armed as soon as Record has been activated and starts at the set time for the set length. If no length has been selected, recording continues until the hard disk is full and then stops automatically.

Record device: Here you can select the sound card driver that should be used for recording. If there is no entry here, or a wrong one, then the card is not installed in Windows correctly. Use the "Info" button to find out the recording properties of the recording device.

File path: Specify the path to the files where the data should be recorded. The yellow folder button opens a query for selecting the path and file name.

Record standard output: Enable this option to record the standard output of the operating system, such as DVD sound, the browser or game audio playback. An actual digital recording is implemented, which means that the exact digital data delivered to the sound card driver by the playback software (e.g. player in browser) is stored.

To prevent this option from being inadvertently left active, it must be explicitly activated again for each such capture process.

Note: Some audio devices integrated into the mainboard also offer the entire audio output as a recording device, often called "Stereo Mix" or "What You Hear". This also works, but the result of such a recording is not a real digital recording, because the analog output signal is fed back to the analog input in the audio device, so a double conversion - from digital to analog and back - takes place.

Tip: Since this recording really records everything that is played back in the operating system, you should disable system sounds, website notifications and the like.

Further Recording Options

Monitoring Mode:

Detailed information about monitoring can be found in "System Options" > "Monitoring Setup" (view page 72).

Monitor: Use this button to activate the LED control displays. Please note that the recording device must be selected prior to this. During recording the LED level displays move slower, but still show each maximum level.

Play while recording: Activate synchronous recording and playing here if this is supported by your sound card.

Note: If you want to record and play over multiple sound cards, small differences may result during playback of longer passages. This is as a result of sample rates of cards which are not 100% synchronous. Ideally you should use the same card for recording as you do for playback if possible. If your card creates a delay between the start of recording and the start of playback, you can balance it out in the "Record Offset" field. To do so play a sample with a noticeable impulse and record it using a loop from the sound card's output to the sound card's input. Then zoom into the Arranger to such a depth that you can precisely pinpoint the offset value.

Create VIP object: Specify via this mode whether or not all recordings should be integrated automatically into a virtual project. The created objects are named with the specified object names.

Update object during recording: This object updates the graphical display of the object while recording.

Read CD/DAT marker: DAT devices and several professional CD players output digital marker information via an SPDIF output e. g. CD track markers or DAT markers). With this recording option all marker information is read from the soundcard's SPDIF input, provided the selected audio device supports it.

Pre-recording (ASIO only): This recording option function inserts audio material that you have added at the beginning of the recording to the beginning of the current recording.

More information about pre-recording is available in "Program Preferences > System/Audio > Record (view page 606)".

Recording offset (samples): If recordings exhibit a constant, undesired shift in relation to the existing audio material in the arrangement, set an offset here which can be used for positioning all recordings.

Advanced options: Detailed information about advanced options is available in File > Program Preferences > System/Audio > Record (view page 606)".

Recording information: This area provides information about the last recording that was made, i.e. **Recording time** (length of the recording), **Remaining time** (remaining recording time), and **Drive space** (remaining storage space on the hard disk being used).

Visualizer: Opens or closes the visualization window.

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

Record: Click this button to start the actual recording.

Note: Please note that during active external synchronization as a "Slave", recording does not start immediately, but rather when the master starts.

Stop: Ends a running recording process (keyboard shortcut: S)

Recording pause: Use this button to interrupt the recording process. The playback marker will continue moving. Clicking "Recording pause" again restarts the recording from wherever you want it to continue.

Take +: Each object or take number (Take 1, Take 2...) is given a marker so that it can be found again quickly.

The Take Manager (view page 189) is ideal for managing the individual takes.

Marker 1: While recording, the option is available to select a marker at the current position of the playback marker, e. g. to mark a take with errors for correction later on.

Close: Use this button to exit the recording options window.

Monitoring

Please also read the chapter "System options -> Global audio options -> Monitoring settings (view page 72)" for more details.

Move Play Cursor

to Beginning

This option positions the playback marker at the beginning of the project.

Shortcut:	Home

to End

This option positions the playback marker at the end of the project.

Shortcut:	End

to Range Start

Sets the start position of the playback marker at the beginning of the currently selected range.

to Range End

Positions the start position of the playback marker at the end of the currently selected range.

to Start of Section

This option positions the playback marker at the beginning of the visible section.

Left/Right Move in Page/Scroll Mode

Use these commands to move the playback marker to the right or left. In "Page" mode, the playback marker moves across the screen to the end of the visible section; at that point, the window view shifts to the start of the following section. In "Scroll" mode, the playback marker jumps to the middle of the section and remains there as the project moves through it.

Keyboard shortcuts:

Movement in "Page" mode:	Right arrow/left arrow
Movement in "Scroll" mode:	Alt + left/right arrow

Note: Once a range open is open, use the "left/right arrow" keys to control the beginning of the range.

Object Border Left

The playback marker jumps to the left to the next object edge in the selected track.

Shortcut: Ctrl + Q

Object Border Right

The playback marker jumps forward to the right to the next object edge in the selected track.

Shortcut: Ctrl + W

Marker Left

The playback marker jumps to the left to the next marker in the arranger window.

Shortcut: Alt + Q

Marker Right

The playback marker jumps forward to the right to the next marker in the arranger window.

Shortcut: Alt + W

Recall Last Stop Position

If the option "Stop at current position" has not been selected in "System options -> Playback", then the playback marker may be positioned at the last stop position with this command nevertheless.

Shortcut: Ctrl + Alt + ,

Repeat Last Position(s)

This command jumps the playback marker to a maximum of 5 previous stop positions.

Shortcut: Backspace

To peak value of all selected objects

When this command is used the playback marker jumps to the position with the highest peak value.

Markers

You will find a number of functions for managing markers. The function keys and numbers can be used to easily save, name, and reopen marker positions.

Detailed information on working on ranges and markers can be found in the "Functions in the project window" chapter.

Marker with Name...

With the option "**Marker with name...**" you can add markers and name them.

Shortcut: ?

Markers with Automatic Numbering

The command "**Markers with automatic numbering**" automatically creates markers with sequential numbers.

Keyboard Shortcut Shift + #

Place Marker at Recording Position

The function "**Marker on record position**" sets a marker at the current recording position during each recording.

Shortcut: Alt + ?

Defined markers are visible at the top border of the project window and can be moved with the mouse. The mouse pointer changes to a double arrow (<->).

Set markers 1 - 10

You can set **markers** at the current playback marker position by selecting a number between one and ten from the menu and entering the corresponding keyboard shortcut.

Shortcut: Shift + 1 ... 0

Jump to markers 1- 10

You can move the playback marker to **marker positions 1-10** by selecting 1 - 10 from the menu and using the corresponding keyboard shortcut.

Keyboard Shortcut: 1 ... 0

Marker on Range Borders

Use this function to set a start and end marker at the borders of a selected range.

Set Markers on Silence

This function automatically sets markers at positions of selected low-volume or silent audio objects. Open the dialog to specify a threshold and a minimum time for detection.

Min. time (ms): Set the minimum time threshold that must be exceeded in order to set a marker.

Threshold (dB): Specify the threshold in decibels here.

> **Note**: When working in destructive wave editing mode, the height of the range in the wave project is set to the level of the threshold to make it easier to see.

Start number: Enter the marker number that is counted upwards to "Marker after silence" here.

Prefix: Enter additional characters/letters which will precede the "Markers after silence" here. This makes it possible to easily differentiate them from markers that may already exist.

Add time (ms): Enter the time range in order to move the "Markers after silence" forward here. The selected object is moved forward by exactly the amount of the time range and filled with silence.

Delete all markers with prefix: Deletes all markers that have prefixes.

Delete all markers: Deletes all markers in the project

Delete VIP objects: This option cuts the object at the set value and may then be deleted.

Rename Marker

If a marker is selected in the marker list (by clicking it), then this function may be used to rename it.

Delete Marker

This deletes the marker at the current position when selected from the marker list by clicking its front edge.

Delete All markers

This function deletes all markers in the active project. The audio markers, however, remain in the wave objects.

Delete Markers in Range

All the markers in an activated range are deleted.

Set New Audio Marker

This command sets an audio marker at the current playback marker position of a selected audio object. If a marker was previously set at this position, then its name will be applied as an audio marker.

Copy Audio Marker to VIP Marker

This copies all available audio markers of the selected object to the marker bar in the arranger window. Application of the VIP markers only occurs within the object boundaries.

Copy VIP Marker to Audio Marker

This copies existing markers from the marker list inside the object borders to the corresponding position of the audio object in the wave project.

Audio marker manager

Audio markers are linked to the audio material, and are visible on the upper side of´an audio object. The purpose of the audio marker is to mark positions within the audio material, this marking will therefore remain regardless of the positioning in the virtual project.

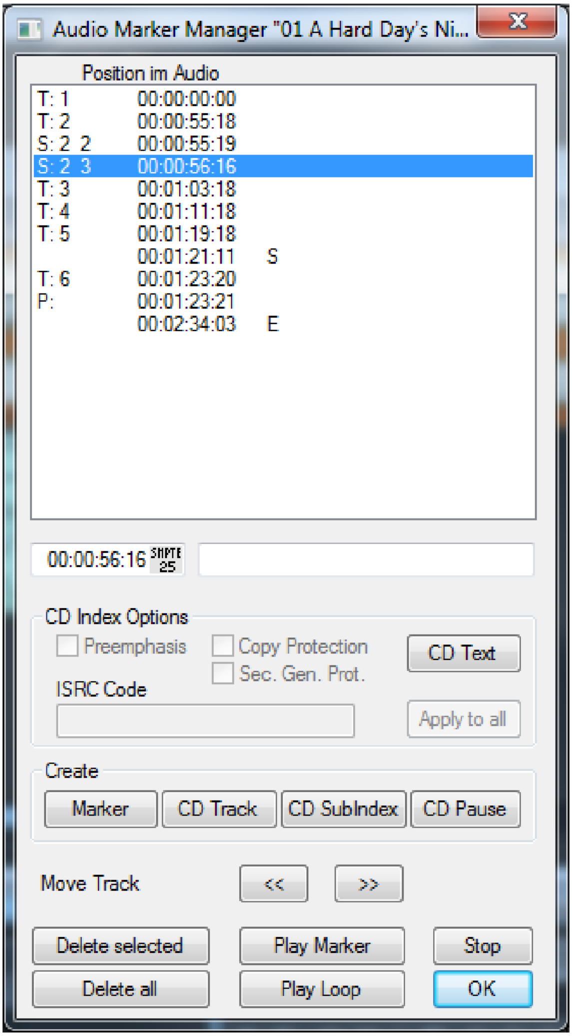
Make Audio markers visible by marking the "Audio markers" checkbox in the view options ("Shift + Tab") in the "Objects" section.

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

The following markers are able to be created and edited via the audio marker manager:

- By right clicking under the upper edge of an audio object in the virtual project a menu will appear, in which all audio markers are listed. You can go straight to them by clicking. In the same menu you can place new Audio markers at the play cursor's position or open the Audio marker manager.

- The position of the Audio marker selected in the Audio marker manager can be changed in the editing field. If you click there a double ended arrow will appear at the selected point, with this you can change the corresponding value. The value content can be chosen in the field to the right.



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

Set Start Marker

This function sets the punch start marker (punch-in) to the beginning of a selected range.

Set End Marker

This command sets the punch end marker (punch-out) at the end of a selected range.

Delete Punch Markers

This command deletes the punch-in marker and the punch-out marker from the marker bar.

Additional Punch Start Marker

Adds another punch in marker.

Additional Punch End Marker

Adds another punch-out marker.

Delete Additional Punch Markers

This command deletes additional punch markers from the VIP.

Insert Tempo Change

Detailed information about setting tempo markers can be found in the chapter „Tempo Editing“ > „Tempo Map Dialog“ (view page 407).

Insert Time Signature Change

Detailed information about setting bar markers can be found in the chapter „Tempo Editing“ > „Tempo Map Dialog“ (view page 407).

Insert Grid Position Marker

Detailed information about setting grid position markers can be found in the chapter „Tempo Editing“ > „Grid Position Marker“ (view page 407).

Marker manager

The marker manager lists all of the markers contained in the current project and makes it possible to jump directly from the list to them or to play them.

To view the marker manager, click the "Manager" button in the toolbar and select the "Markers" tab.

Shortcut: Ctrl + Shift + Alt + M

Detailed information about this is can be found in the chapter "Manager -> Marker Manager (view page 184)".

Auto JamSession

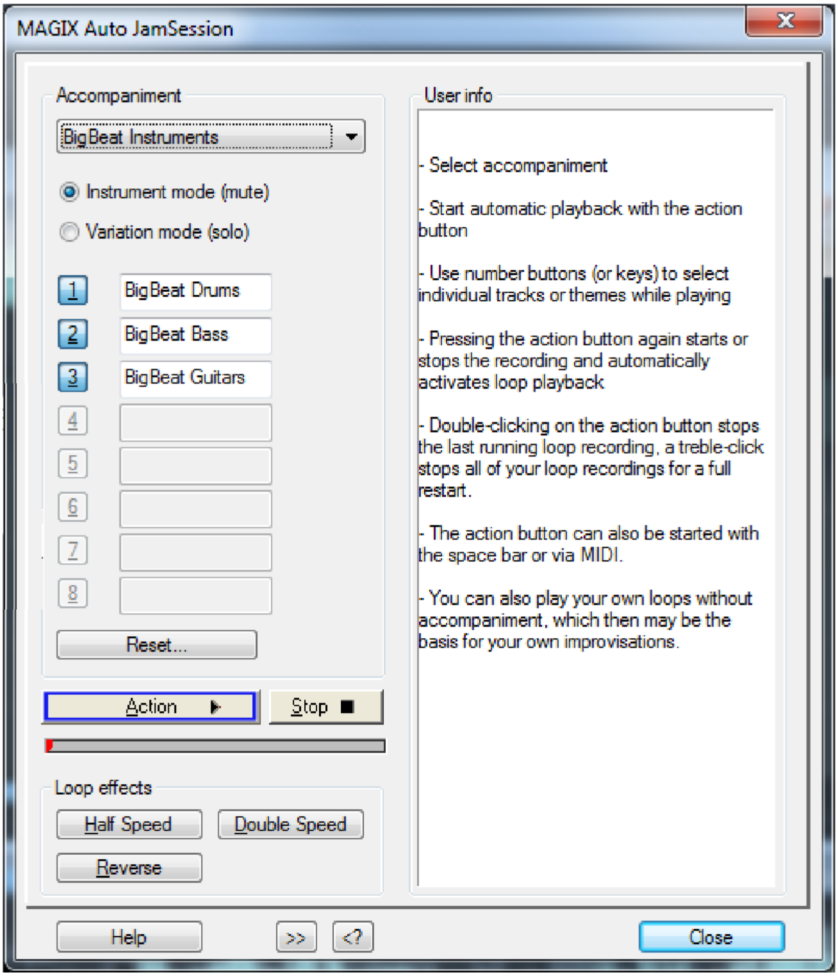
You can now jam quickly and creatively with your virtual band thanks to Auto Jam Session's practical loop automation function. Record your instrument on separate tracks and develop the song structure systematically.

The recorded tracks are then put straight into an endless loop after the recording.

Auto JamSession - access

Open an Auto Jam Session via "Playback -> Auto JamSession..." or via the "Record" mode box in the transport window.

Shortcut: Shift + J



Auto JamSession – modes

There are numerous ways to record a song:

- If you want to play loops yourself, leave the "accompaniment" field empty. The loop that was recorded first is used as set the tempo of the song in BPM (beats per minute). The buttons 1-8 function as mute or solo buttons depending on their set modes.
- If you would like to jam to another prerecorded session, select one of the included accompaniments. The tempo of the song will be determined from the template. There are two types of templates:

Template in instrument mode (mute): The accompaniments consist of multiple instruments that can be muted using buttons 1-8 (or the corresponding number keys on the keyboard).

Variations mode (solo): These accompaniments contain multiple variations of the same instrument that you can switch back and forth during playback via buttons 1-8.

Reset: This function allows you to switch between various "Reset" functions. You may either:

- set the "Mute/solo" buttons back to original state or
- stop the last recorded loop at the playback marker position or
- stop all imported loops at the playback marker position or
- stop all loops at the playback marker position or
- delete the entire project.

Auto JamSession - operation

After launching, you'll see the simple view of the Auto JamSession. On the right-hand side, you'll find useful hints on how to use the feature.

"Action" button: The "Action" button on the lower left is an important control element, and it may also be activated via the spacebar. Repeated clicking triggers the various functions in series to produce your own jam session.

1st click: Playback begins. Recording does not start.

2nd click: The first recording starts. Record your first beat(s). As a symbol that recording is running, the "Action" symbol turns red.

3. Click: Interrupts the recording while playback continues. The portion played is inserted into the arrangement as a looped object.

4th click: In the next track, the recording starts. You can also hear the previously recorded loop.

5th click: The recording is interrupted, playback of the first and the newly recorded loop continues.

etc.

This allows any number of additional voices to be played.

Double clicking the "Action" button stops the most recently recorded loop and cuts it at the current playback marker position. **Triple clicking** cuts all played loops at the current position.

Clicking the "Stop" button ends the playback/recording immediately.

Auto JamSession - loop effects

You can change your arrangement with these real-time effects. The loop effects influence either every track or just the last track recorded (see below).

From a technical standpoint, these loop effects belong in the object effects category. If the loop effects are activated, separate objects will be created from the loops.

Half speed: Press this button to halve the playback speed.

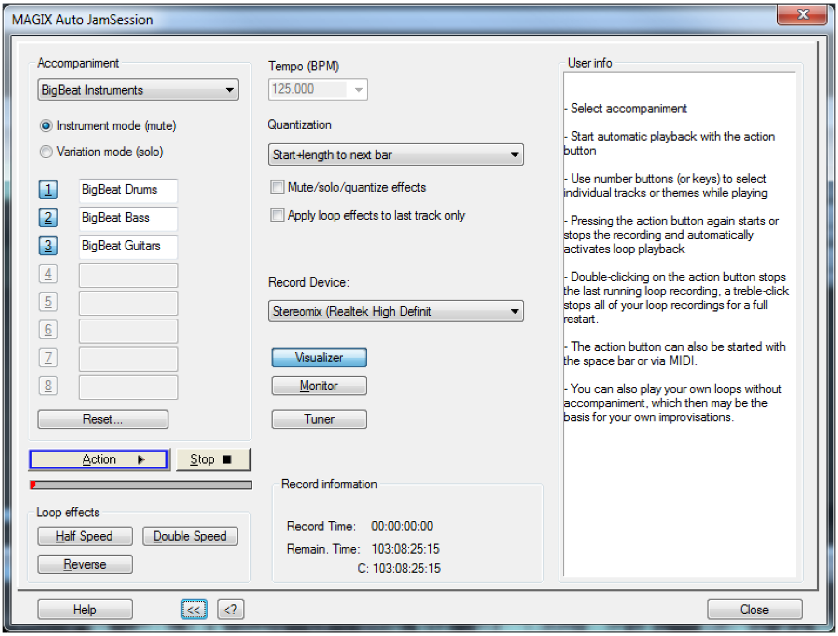
Double speed: Press this button to double the playback speed.

For example, "Half speed" allows you to play nearly impossible guitar solos. Switch the existing backing arrangement to "Half speed", record your guitar solo, and then click "Double speed" again. Your accompaniment plays in original speed again and your solo now sounds twice as fast and an octave higher.

Reverse: This effect plays the whole arrangement backwards. Opening it again resets it.

Auto JamSession - advanced dialog

Clicking the ">>" button opens the enhanced Auto JamSession dialog.



Tempo

Select the tempo from the list or enter one of your choice here.

"–" indicates that the speed adapts to the first recorded loop or to that of the accompaniment (if chosen).

Quantization

The quantization ensures that recorded loops fit into the arrangement that is created.

There are multiple options for quantization:

Length of the next whole loop: The new loop is recorded with the same length or a whole-numbered multiple (2x, 3x, 4x...) of the original loop length. The loops always remain synchronous, but do not need to have the same starting point. This is the preset quantization mode.

Start + length of the next whole loop: Recording only starts after the next loop border. The length is quantized like in "Length of the next whole loop". The starting points of the loops are all at the same positions.

Length to previous whole loop: The loop end is shortened to the last exceeded loop border. This is useful in case the "Action" button (spacebar) is only managed to be clicked shortly after the next loop passage during playing.

Start + length to next bar: The starting point of the recording and the loop length are adapted to the bar. This is useful if you have recorded the first loop over multiple bars.

Start + length to the next beat: The starting point of the recording and the loop length adapt to quarter beats. This is useful if very short loops are played.

Both of these last two options are useful if the "Tempo" has been set or is given by the accompaniment.

Freestyle: Quantization is deactivated in this case. No more loop starts or loop lengths are adapted when recording.

Quantize mute/solo/effects: This option also quantizes the control of solo, mute, and other effects (double speed, half speed, reverse).

Only use loop effects on the last track: The loop effects are always used on the last track.

Auto JamSession - record device

You can select the desired sound card or sound card input via this dropdown menu.

Visualizer: This button switches the visualization on or off.

Monitor: This button activates/deactivates monitoring. This means that the incoming signal is displayed in the visualization and played by the sound card output.

Tuner: This button opens an additional visualization in the "tuner" mode for tuning instruments (e.g. a guitar).

Auto JamSession - recording information

Recording time: The recording time indicates how long you've been recording.

Remaining time: This indicates how much longer the hard disk may recorded to. If you have multiple hard disks or hard disk partitions, the recording capacity of each of these are also shown.

Retrospective Recording (MIDI Pre-Recording)

Samplitude creates a MIDI object in the recording-ready MIDI track that may be adjusted in terms of **buffer** length via "**System options -> MIDI**".

Retrospective recording intermittently saves MIDI events and audio signals in an adjustable memory buffer in RAM. This takes place while running, even if a recording is not being made via the record button. This enables creative moments to be captured and then integrated into the project as a file/object. In this case, it doesn't matter whether the project is playing or not.

The MIDI object may be added at the cursor position or synchronous to the last playback. A **prerecording** of 2 seconds is also saved during MIDI recording as an extra take. By changing the takes in the **take manager**, e.g. from "MIDI Take3" to "MIDI Take3 **PreRec**", you can drag the object to the left to restore the rhythm played prior to the actual recording. The object borders will be adjusted to exchange the take.

Note: If you would like to add an **audio recording** from the buffer into the project retroactively, drag the newly created object to the left after recording (according to the **set pre-recording time in the recording dialog of the system options**).

MIDI Record Mode

The MIDI recording modes specify how newly recorded MIDI data is inserted into the virtual project if there are already MIDI objects at the recording position.

Normal

In this recording mode, a new MIDI object is created over the existing object for each recording. All objects are maintained. This way, you can record multiple takes of a passage and then compare them to the object with "Alt + right click".

Overdub

In this mode, the data for each new MIDI recording is mixed with already existing takes.

Replace

In this mode, MIDI files of the existing object will be replaced with newly imported ones. If you record over several objects, these are combined into a single new object.

MIDI Panic – All Notes Off

This command sends a note off command to all MIDI devices that are activated in the MIDI options for all 128 notes on all 16 channels. The sustain (Controller 64) is also shut off, and the pitch wheel and modulation are reset to 0. All VSTis used in the project will also receive the all notes off command.

If MIDI tracks or objects exist in the project, the same function is also accessible by clicking the stop button in the transport console or toolbar when the project is stopped.

Automation menu

The content of the "Automation" menu corresponds to the Automation context menu of the selected track. For a description of the menu items, see chapter "Automation", section Automation context menu (view page 443).

No effect (Track)

If you select this display option the track automations are deactivated.

Edit selected curve

Curve generator

This menu item opens the curve generator.

Invert

This command inverts the activated automation curves.

Thin Out

This function "thins out" the activated automation curve by reducing the number of automation events. During recording, the automation events are placed in very short intervals. The "Thin out" command reduces the number of curve points. The course of the automation is then displayed and reproduced more accurately.

Thin out automatically

The command "Thin out" is executed automatically after automation data is recorded or drawn.

Transfer Track Automation to Object

You can use this function to transfer volume and panorama automation to objects.

Convert object automation into track automation

With this function you can transfer the volume and panorama automation of objects to tracks.

Inactive

Use this command to deactivate the selected automation curve.

Curve Color...

This command displays a color palette to specify the color of the active curve.

Logarithmic

This function switches between the linear and the logarithmic display of the automation curve.

Copy, Paste, Delete

The active automation curve may also be copied, pasted into other tracks, and deleted.

Delete All Curves

This command deletes all automation curves in the corresponding track.

Automation Mode

Specify the Automation mode here.

> You can find detailed information in "Automation modes (view page 430)".

Hide Automation

Use this option to deactivate the display of automation curves for all tracks.

Show Track Automation

Select this display option to display only the track automation.

Show Object Automation

If you select this display option, only the object automation will be displayed for all tracks.

Show Only Selected Curve

Views only the selected curves for the corresponding track. This helps to maintain a clear overview if several automation curves have been created.

Show all curves (not selectable)

The deselected automation curves are grayed-out. They cannot be activated with the mouse.

Show all curves (selectable)

The deselected curves are also shaded, but may now be activated with the mouse.

Show lanes for all curves

This opens the automation lanes and corresponding automation lanes are created for all existing automation curves.

Select previous curve

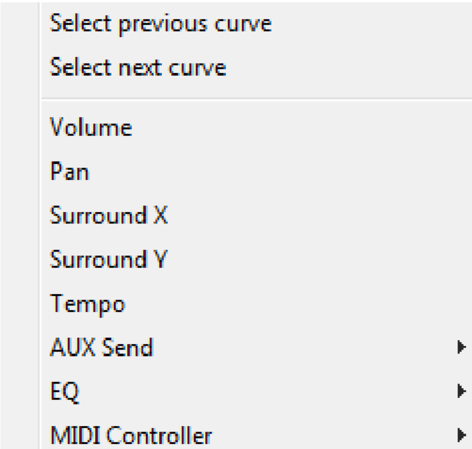
With this you can select the last edited curve.

Select Next Curve

With this you can call up the next curve from the selection list. The last section of the context menu lists the ranges for which automation parameters are available during automation of the corresponding track.

Select Automation Parameters

In the lower section of the "Automation" menu you can select various automation parameters and then control them with the automation controller in the track editor.



Effects Menu

This menu entry provides fast access to the relevant real-time effects at object level. Correspondingly, menu entries for all object effects are provided here.

Note: All effects opened using this menu are calculated destructively, provided the option "**Apply effects offline**" is active. The option to work with a copy is available in order to preserve the original audio material. The "**Create copy**" option is already selected in the corresponding dialog.

Effect Dialog Bar

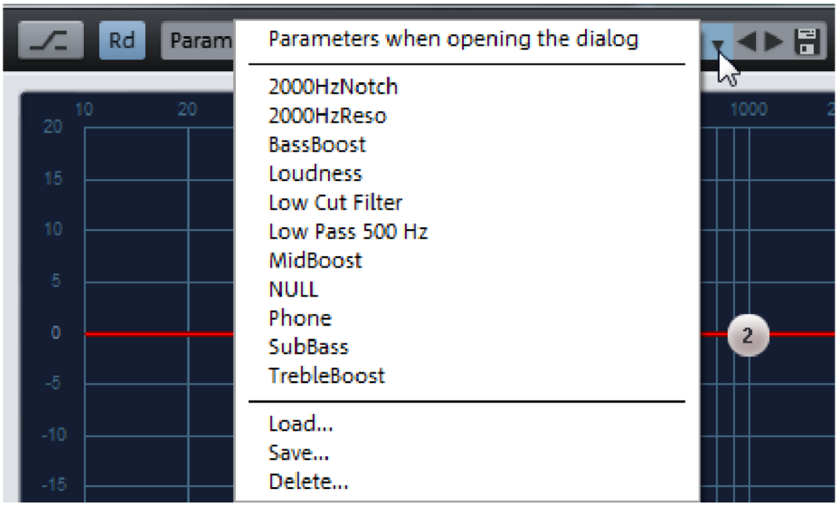
In several effect dialogs you have direct access to the automation button which also provides access to the automation menu. This is made possible by a newly created header in the respective dialogs.

In the presets input field you can load, save or delete settings.

By selecting "Parameters when dialog is opened", you will undo all changes made in the dialog since it was opened.

By closing the dialog, you will apply all current settings.

Clicking on the downward pointing arrow, you will see all presets at a glance. Below the preset list you can open dialogs for loading additional presets and for deleting selected presets. You can navigate through presets using the two arrows pointing in horizontal directions. Current settings are saved to disk.



Here is an overview of the other control elements:

Bypass: The effect is removed from the signal route. This enables the unedited signal to be easily compared to the result of the current effect settings.

Mode: By clicking on this button you can toggle between the selected automation write/read mode. By right-clicking you can get to the Automation context menu.

A/B comparison:This control element is only displayed when it is supported by an effect. You can copy each selected setting using the arrow symbol to a different location. This way you can experiment with successful settings without losing them.

Reset: This control element is only displayed when it is supported by the respective effect. All parameters are reset to default settings.

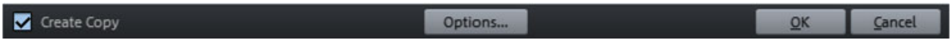
Play button.

Solo button. This control element is only displayed when it is supported by the respective effect. The Solo mode will be activated for the corresponding effect.

Help: You can get extra information in the help section.

Notes concerning offline effects editing

- When editing objects via the context menu item "**Edit audio file**", offline effects always have an effect on all of the objects that reference the same audio material. If this is not desired, access the command "**Edit a copy of the audio file**" prior to destructive editing via the selected object's context menu. Samplitude will create a copy of the audio material and add this to the project folder. The corresponding object now refers to the copy that has been created. Following this, offline effects may be applied to the selected object without influencing other objects.
- Activate the option "**Create copy**" in the offline effects dialog to use the undo function later (this allows you to undo destructive editing).



- The following functions are available exclusively for audio files: "Sample Manipulation -> Sample Amount *2, /2", "Show/Hide", "To Zero" and "Build Physical Loop".

Note on the "Undo" function: Please tick "Undo active" via the "Options menu -> Program Preferences -> Undo definitions (view page 624)" for virtual projects and for audio files in order to activate the undo function.

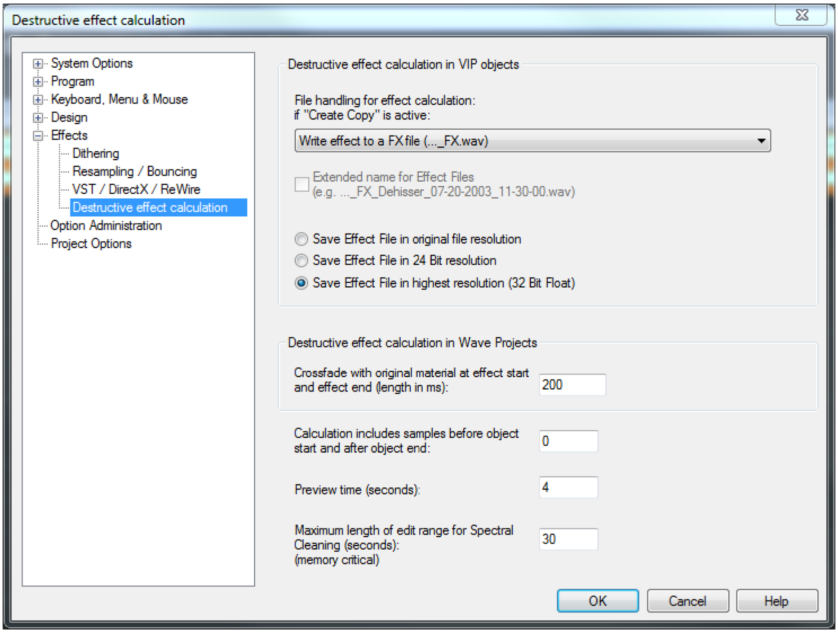
- Note that most functions only work within the selected range in the wave window. To use the functions on the entire audio file, the entire file must be selected (keyboard shortcut: "A"; menu item: "Range -> Range all").
- On the other hand, selected objects in VIP projects are changed independently of the selected range across the entire range (the only exception here is the "Get noise sample" function. (The "Get noise sample" function is an exception). If you do not wish to change the whole object, spit it at the range edges using the "T" key and activate "Auto-crossfade" mode to avoid crackling transitions.

OK: The edit is applied to the selected range of the sample or the selected object.

Cancel: Click this button to close the dialog window without applying any editing.

Advanced options for offline effect calculation

These settings may be opened in the corresponding effects dialog by clicking the "**Advanced options**" button. Alternatively, this dialog may be accessed via "File -> Program Preferences -> Destructive effect calculation...".



In this dialog, choose to write a copy with offline effects from the many options.

Destructive Effect Calculation in Wave Projects

If this is not desired, access the command "**Edit copy of wave project**" prior to destructive editing via the selected object's context menu. Samplitude will create a copy of the audio material and add this to the project folder. The selected object now refers to the copy that has been created. Following this, destructive effects may be applied to the selected objects without influencing other objects.

The dropdown menu of the dialog provides selection between three styles of saving when executing offline effects. Saving is required in order to be able to undo offline effects. A check must be placed beside "**Create copy**" in the effect dialogs for this to be available.

Note: Deselect this function if you're sure that the undo function will not be required. A lot of time and hard disk capacity can be saved if a copy of the wave is not created.

- **Append effect to original file**: The object with the calculated effect is added to the original file. However, this has some disadvantages: When working in an integer wave file, the effect is also saved at this resolution, which is not always desirable when working with 32-bit resolution. The length of the wave file is also changed, which may cause problems when working with looped objects or in various VIPs.
- **Write effect to an FX file (._FX.wav)**: The result of the effect calculation is written to a separate file with the extension "_FX.wav" so that the original wave project remains untouched. This also makes it possible to execute the effect calculation in the 32-bit float format, to save it, and thereby keeping the full quality of the effect. The file is not overwritten with every new effect calculation, but rather the new effect is simply added to the effect file.
- **Generate a new FX file for each effect**: All offline effects are saved in separate files with sequential numbers. Alternatively, these files may be given detailed names featuring effect descriptions and the date.

The effects file may also be saved in the format of the output file, in a 24-bit format, or in a 32-bit float format.

Offline effects calculation in wave projects

When opening a offline effect in wave projects, the result of the effect calculation is always added to the opened wave project. The options of the offline effect calculation on VIP objects described above are not taken into account.

Temporary files for the "Undo" function are only created with offline effect calculation if the "Undo" function for wave projects is also activated (Shortcut: "Y" – Program -> Undo) and "**Create copy**" is ticked in the corresponding effect dialog.

If an effect is applied to a specific range in the wave project, a crossfade may be inserted between the effect and the original at the beginning and end of this range. Enter the **length of the crossfade in milliseconds**.

Moreover, you can calculate a number of **additional samples before object start and after object end**.

Preview Time (seconds): Preview time is the duration calculated for listening to effects in the preview function.

Maximum length of editing range for Spectral Cleaning (seconds): Enter how long (maximum) the audio material should be edited during spectral cleaning.

Amplitude

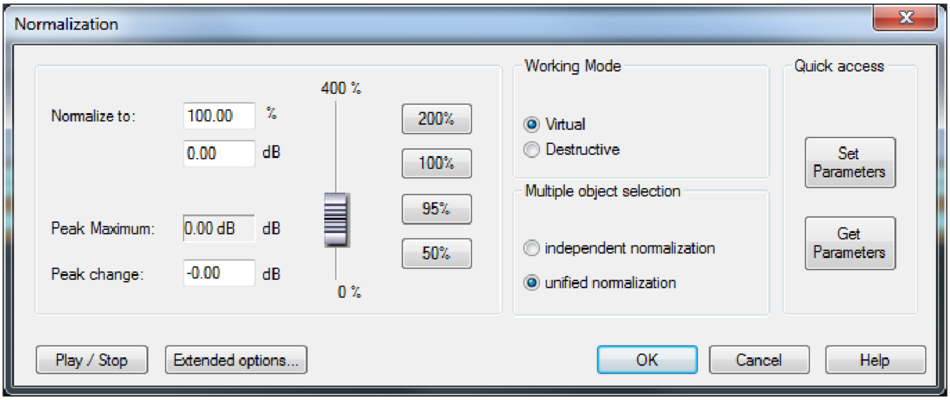
Normalize

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

Use this function to fully modulate samples or overmodulate targeted audio segments. Note that the noise level also rises during normalizing.

This function is especially useful when converting from high to lower bit rates. This guarantees that the otherwise low dynamic range is used to its full extent at lower rates.

Shortcut: Shift + N



Note: If you experience very slight clipping during recording and then proceed to normalize the material, you won't achieve the same quality as when you produce an overmodulated recording! For example, if you only modulate half of the material, then your recording will have a quality of 15-bit samples – normalizing to 100% will not change anything.

Normalize to: Set the value which the audio material should be normalized to by entering it into the input field, moving the fader, or selecting one of the presets (50, 95, 100, or 200%). The value will be shown in % and dB (100% = 0 dB = max.). Values above 0 dB will cause digital clipping.

Level maximum: This indicates the highest resulting volume for the selected range or object in the corresponding setting.

Level change: Displays the level change in dB according to the selected normalize level and the detected level maximum.

Working Mode

Virtual: This function performs real-time normalizing on the selected objects. In this case, the sample data is not changed, but rather the object volume is adjusted so that the loudest part of the object reaches the selected normalization level. This virtual normalization is non-destructive, unlike destructive normalization, which changes the audio material irrevocably. The object to may be restored to its original volume setting by selecting "Reset", which is located below the volume fader in the object editor.

Destructive: The audio material is physically altered. A tick is set beside "Create copy" in the dialog.

Multiple Object Selection

When multiple objects are selected, there a number of different techniques to normalize them.

Independent normalization: Each object is normalized according to its own maximum level.

Uniform normalization: The maximum level is detected from all objects and each object is normalized according to that value. This is active by default.

Quick Access

Set parameters: Use this button to make the settings of the normalization function accessible via quick access.

Get parameters: Use this button to get the current quick access parameters in the dialog.

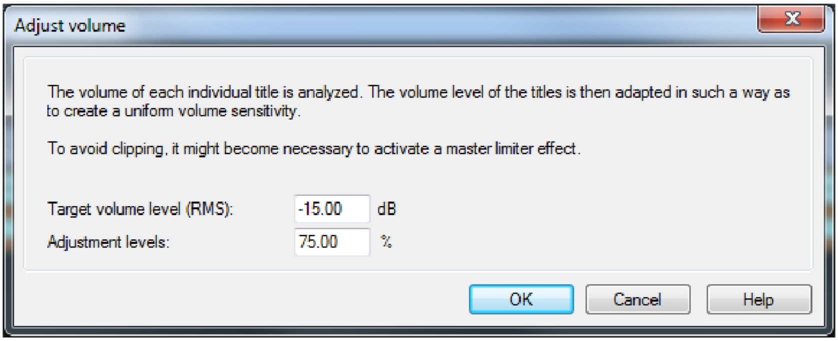
Normalize (Quick Access)

This function enables the specified settings in the "Normalize" dialog under "Quick access -> Set parameters" to be used immediately.

Shortcut:	N

Loudness Adjustment

This command executes an automatic RMS normalization. The loudness of each individual track object is analyzed and the level of the tracks is adjusted to create a uniform volume sensitivity. The preset target value is -15 dBfs. The target value can also be adjusted manually. The closer the value is to 0 dBfs, the louder the signal will be.



The adjustment intensity allows you to determine in percent how precisely the track will be matched to the target loudness value.

Fade In/Out...

Use this function to show or hide selected ranges of samples in a wave project. The amplitude will change according to the progression from the start value at the beginning of the range to the end value and the end of the range.

A simple fade-in process may be activated in the dialog via the parameters "Fade start in % = 0", "Fade end in % = 100". A simple fade-out may be executed by setting "Fade start in % = 100", "Fade end in % = 0".

You can have the transition display in three different curve envelopes: exponential, linear, or logarithmic.

> Note that in virtual projects, powerful real-time fade and crossfade options are available that usually make the use of destructive fade routines unnecessary.

Set Zero

The level in the selected range is set to zero. This allows unwanted parts within a sample to be removed.

Dynamics Effects

What compressors are available in Samplitude?

There are four different modules for dynamic editing available in Samplitude:

1. **Essential FX Compressor (view page 857):** A simple but effective tool for reducing dynamics with soft curve characteristics and an adaptive regulation process. The "essentialFX Compressor" is a very musical compressor.

2. **Advanced Dynamics (view page 763):** This module is a very comprehensive tool. It is a combination of a classic dynamics module (compressor/expander/gate) and an approximating limiter. The result is an undistorted, optimally modulated signal with defined volume. Setting the parameter is done either by direct input or by graphically changing the characteristic. Level detection can be derived in either "**Peak**", "**RMS**", or "**Fast**" mode.

3. **Multiband Dynamics (view page 769):** This module offers extensive dynamics editing options. The advantage of dynamics manipulation across **multiple frequency bands** versus a standard process is that the danger of pumping and other side-effects sinks drastically. For instance, it can prevent a level peak in the bass range from overpowering the entire signal. Multiband technology also lets you specifically edit individual frequency ranges.

4. Samplitude's **AM-Track**, **AM-Pulse**, **AM-Phibia**, and **AM-Munition** also provide specific dynamics editing functions that simulate analog effects processors. Read more about this in the extensive chapter "MAGIX plug-ins (view page 849)".

5. **sMax11 (view page 778):** With this module you can increase the volume of the audio signal by setting an input amplification (Gain-In). Essentially, this involves a hard or brickwall limiter with input amplification.

6. Essential FX Limiter (view page 861): This plug-in is a simple yet effective tool for increasing the loudness of your audio signal. This creates a compressed but loud signal without allowing the defined output volume to be exceeded.

Which compressor should be used and when?

When to use a certain dynamics module depends on how specific the change needs to be. The "essential FX Compressor" module works well on mixer channels to quickly and efficiently influence the dynamics.

However, the sound of an object should normally be changed very specifically. In this case, the dynamics of an instrument can be varied between the verse and the chorus depending on the playback method. This is a desired relation that can be preserved or even forced using precise compressor settings. We recommend using "Advanced Dynamics" to change the dynamic behavior of the source material as optimally as possible.

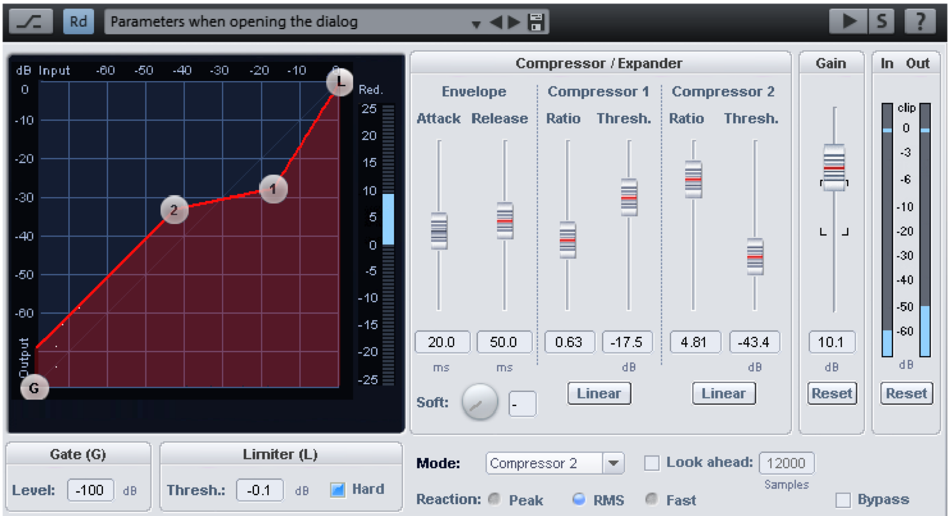
Experiment with "AM-Track", "AM-Pulse", "AM-Phibia", and "AM-Munition" to add characteristic pre-amps or tube sounds to your arrangements.

The master module in the mixer features "Multiband Dynamics" to give the entire mix its final balance. Because an increase in volume using a multiband compressor can lead to clipping, it is necessary to use a limiter to suppress the clipping samples.

We recommend using "Multiband Dynamics" for each object/CD track when mastering CDs. This enables you to react to the differences of CD tracks with different multiband dynamics settings.

The "sMax and the "Essential FX Limiter" are maximizer/limiters that increase the loudness of the audio signal.

Advanced Dynamics

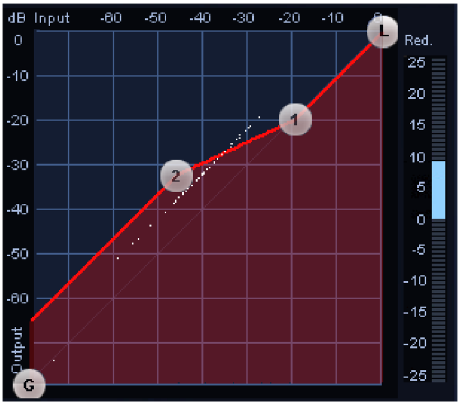


Advanced Dynamics - general functional elements and presets

Save, **load** and **delete** functions are integrated into the presets list, where they are available in the lower section. The default file extension is ***.dy2**.

You can also load all dynamics presets (*.dyn). The parameters are converted to those available in the "Advanced dynamics" module. If saved from within the Advanced Dynamics processor, the preset will be saved with the *.dy2 extension.

Dynamic scope: When you play the selected object/sample, you'll see the in and out level of the signal as a white, scattered line.



Reduction display: To the right of the graphic display you'll see the level reduction of the played audio material.

In/Out displays: These meters show the input and output displays in dB.

Reset: Reset the reduction, in and out displays.

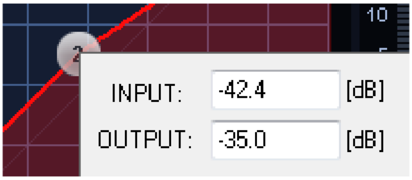
Mode: Modes are typical applications of "Advanced Dynamics". They constitute default settings that makes the graphical editing of the reference line easier. For this, the number of editable parameters are limited by the choice of mode, so that e.g. with pure limiter application, the other curve bases no longer change. The following modes are available to you:

- Simple compressor
- Two-point compressor
- Expander
- Gate & limiter
- Limiter

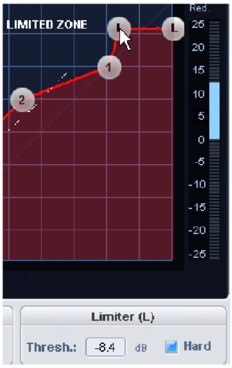
Advanced Dynamics - editing of reference curves (graphically)

To graphically edit the static transfer characteristic, press the left mouse key in the reference line field on one of the curve bases and move it with the key held down. Moving the base points at one end influences the other points of the corresponding parameters.

In order to directly position the point in the reference line field, press the right mouse key. A small dialog appears, where you can read and write your input and output values.

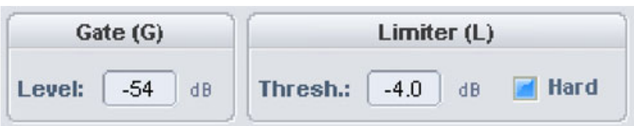


If you turn on the "Hard" parameter in the limiter and drag the curve point for the limiter threshold down, a "Limited zone" label will appear for the area below the limiter threshold. The limiter pushes the levels below the "Limited zone".



To hide a handle, adjust the corresponding threshold to 0dB using the slider; the reference line will not change in this case. In order to make it visible again, the threshold must be below the limiter threshold.

Advanced Dynamics - editing the reference line (parametric)



Gate (G)

Level [in dB] determines the minimum level at the input. All signals below this level will have an output signal level of 0 dB.

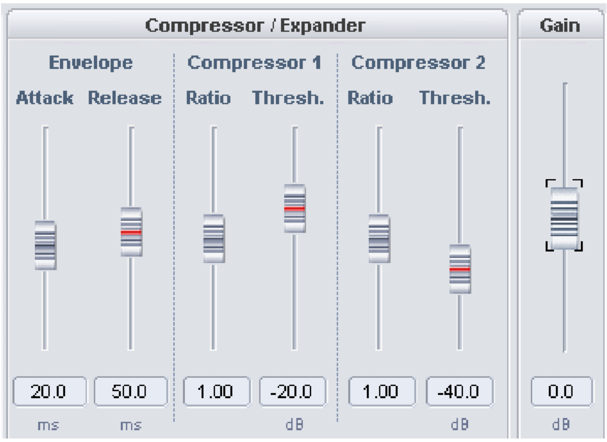
Limiter (L)

Level [in dB] determines the maximum level at the output. Note the "Hard" setting.

Hard: When you switch on this parameter, the output signal limits exactly to the set threshold value. This means that no sample lies above the set limiter level. Of course, the signal is not simply cut off at this limit. The used algorithm brings the signal as close as possible to the limit without changing the original sound. If the "Hard" setting is not active, the algorithm only controls according to the set transfer reference line and behaves like an analog limiter.

Soft: This parameter specifies the curve of the reference line in values from 0 to 20, whereby 0 means no curve. Curving is useful if the continuous change between uncompressed and compressed signals becomes apparent, i. e. the signal level alternates around the inflection point. With "Soft", a smooth transition is achieved.

Advanced Dynamics - envelope



The time constants affect the sound characteristic significantly. In this way, specific time constants may lead to distortion or "pumping" effects.

Attack: Here you can set the time frame between the crossing of the threshold and the maximum extent of the effect in milliseconds.

Release: Here you can set the time frame between the falling off of the threshold and the complete dissipation of the effect in milliseconds.

Compressor 1/2 threshold: Here you can set the input threshold in dB, above which the respective component is activated.

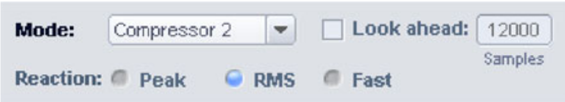
Compressor 1/2 ratio: With this parameter you can control the compression ratio above the threshold. The "Linear" button resets the ratio to 1:1.

Gain: Use this fader to set the whole static reference line in dB.

All changes to these parameters directly influence the static transfer characteristic line. However, you should note that all these parameters influence each other.

Advanced Dynamics - Dynamic parameter

The "**Reaction**" function is applied to the source material similar to the compressor.



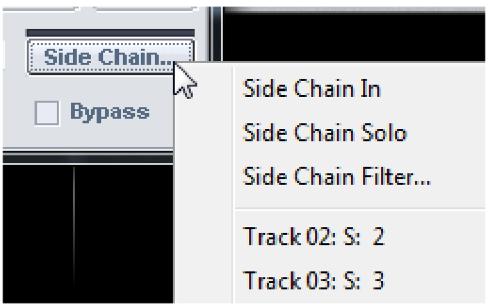
Peak: This mode uses the signal's peaks as a basis for calculation. The dynamics module reacts to level peaks quickly and efficiently.

RMS (Root Mean Square): This mode is involves the signal's average volume, which corresponds with the dynamic behavior of analog dynamics modules. It sounds relatively round and balanced, less focused. The time constant for signal detection of the control signal only refers to the setting of the "Attack" fader. Both time constants are analyzed for signal equalization when manipulating signal dynamics.

Fast: Use this option if the object dynamics should be modified only a little. The maximum level at the output is never higher than the limiter.

Preview: If this switch is activated, the dynamics section works in preview mode. This can help prevent artifacts (pumping) and overmodulations. On the other hand, without a preview, this option will not automatically "level" steep attack phases in the waveform, which may result in sharper-sounding audio material. To emulate the sound of analog dynamic modules, it is recommendable to work without preview. The length of the preview may be entered in the "Samples" field.

Side chain...: If "**Advanced dynamics**" is used as a track or master effect and the track is not a Surround master, the "side chain" button will appear. Enabling this option is used to request an external signal to calculate the internal control signal.



Clicking the "side chain" button opens the side chain menu.

Side chain in: Activates and deactivates the side chain function.

Sidechain sole: With this option you can listen to the sidechain signal. The compressor will be switched to "Bypass", and only the input of the side chain will be played. When closing the plug-in dialog, the "Side chain solo" function is automatically reset. "Sidechain solo" is also ideal for acoustic control when applying the sidechain filter.

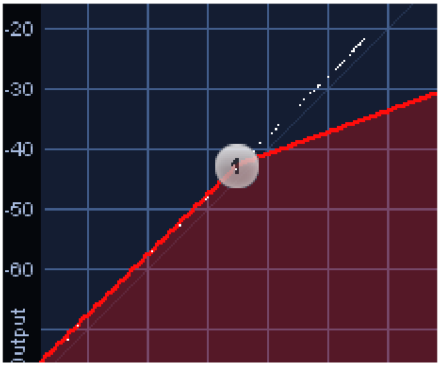
In the lower half of the menu all tracks are listed, of which you can set one or more as sidechain inputs. **In the AUX section of tracks that used as sidechain signals or mixer channels a new AUX send fader will be assigned for the target track.**Internally, a side chain bus (AUX bus) is created. To differentiate from the "normal" AUX buses, a "**>**" is added. "Pre-fader" is set to catch the side chain signal. This means that the sidechain remains independent of the fader position of the sending channel.

Right clicking on the AUX send button selects this as "Side chain send". Thereby you can use further track as the source for the sidechain signal at a later date.

Sidechain filter: The sidechain filter can be filtered by a parametric equalizer.

Advanced dynamics - Dynamic scope

Dynamic scope appears when you play the selected object/sample in an open dialog. You can then see a scattered white line, a "snow cloud" whose individual points represent the input and output values in dB. This way, the change to the samples that is achieved via advanced dynamics is constantly visualized.



You can clearly see the way dynamic scopes work in the following example:

Select "ZERO" and press "Play;, now the effect won't change the signal. The scope shows all samples on red lines and this means this means that all input values equal to the output values. Slide the gain fader upwards to see that the "snow cloud" follows the reference line. If you select very high time constants, this takes place relatively slowly.

Selecting a relatively short attack time (10 ms), a long release time (400 ms) and a severely arched reference line can also move the effect towards "pumping". The "restlessness" of this dynamic scope mirrors the "pumping" graphically.

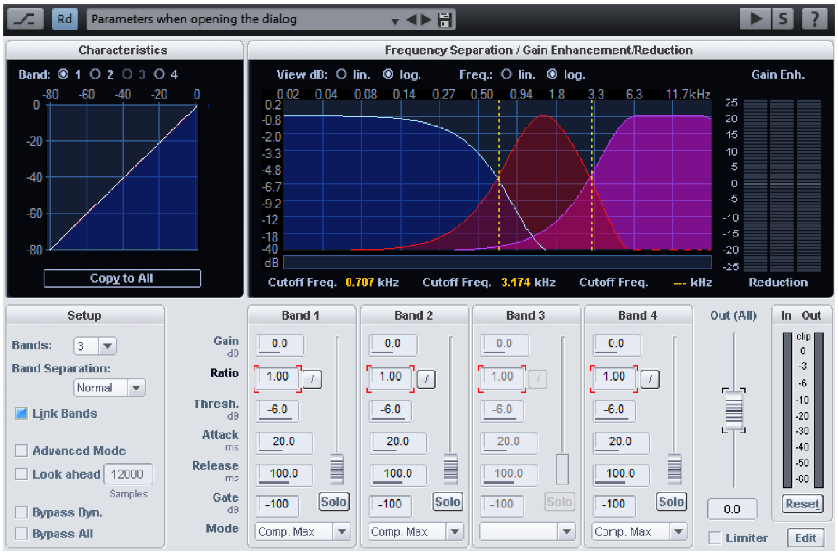
Advanced Dynamics – tips and tricks

- Select one of the provided presets as a general setting. You can now adapt the reference line to the source material with the help of the dynamic scope and very precisely separate the signal components.
- If the scattered line of the dynamic scope moves very jerkily, this is probably because of the settings of the time constants. Set these according to the sound preferences and then save the personalized presets.
- In order to couple an "analog" limiter with the hard limiter algorithm, define the Limiter level at the maximum desired sample level. Once set, an "analog" limiter may be set by using an extreme ratio setting (compressor 10.00 or expander 0.10).

Multiband Dynamics

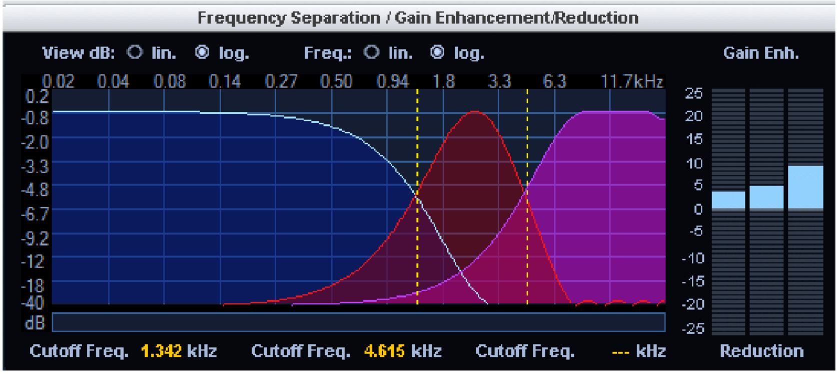
The "Multiband Dynamics" section lets you edit the **dynamics with up to four independent frequency bands**. The entire signal is split into individual frequency bands in this case. The dynamics are then edited separately for each band.

The combination of the individual bands is 100% phase-neutral, and thanks to the FIR Complement Filter, technique no discoloration of the frequency response occurs. In other words: If there is no dynamic editing in the individual bands, i.e. the signal is only split into the bands and then reassembled, the audio material will be perfectly compiled to the state it was before. The advantage of dynamics manipulation across multiple frequency bands versus a standard process is that the danger of pumping and other side-effects sinks drastically. For instance, the function is able to prevent bass peaks from reducing the entire signal. Multiband technology also lets you specifically edit individual frequency ranges.

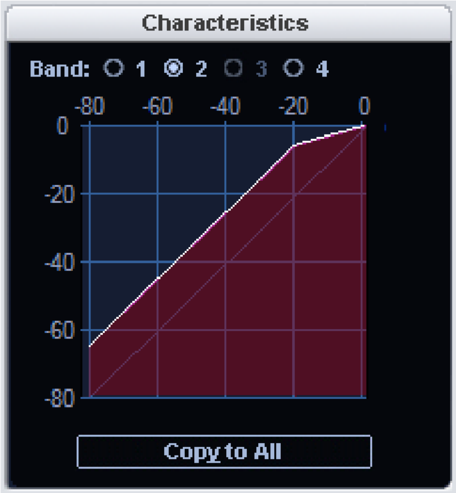


Multiband Dynamics - overview

In the "**Frequency separation / Gain Enhancement/Reduction**" dialog, you'll find the parameters for the filter bank. The graphic display shows the frequency of separate bands. The left axis label shows damping in -dB, the top label shows frequency in kHz. The individual curves are color labeled.



The graphic element to the side shows the **dynamics characteristic curve**. It illustrates the ratio from the input level (upper axis in dB) to the output level (left axis in dB).



The reference curve for the band currently selected is always displayed.

The dynamics section of the individual bands is equipped with the following parameters: "Gain", "Ratio", "Threshold", "Attack", "Release, "Gate Level" as well as the modes "Compr. Max", "Compressor", "Limiter", "Limiter 100", "Expander", and "Gate".

Multiband Dynamics - general controls

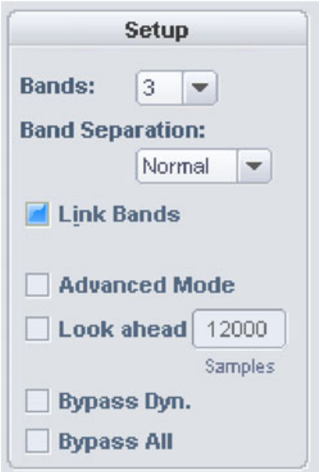
Save, **load** and **delete** functions are integrated into the presets list, where they are available in the lower section. The default file extension is ***.mdy**

Reference line

Band selection: Here you can select a band for dynamic editing. The reference curve of the currently selected band is displayed in white, while the other bands appear as lines in their corresponding color.

Copy to all bands: After pressing this button the parameters of the chosen band, i.e. the parameters whose values appear on the screen, are applied to all bands. If you also have "Link bands" selected, the values for all bands remain coupled to one another.

Setup



Bands: Here you can select the number of bands, between 1 and 4. Set to "1", no separation into different frequencies takes place, and the algorithm works like the standard dynamic section. The algorithm CPU load rises with increasing number of bands.

Band separation: See below

Link bands: When this option is active, the changes to the dynamic parameters affect all bands. Often, it isn't necessary to set dynamic parameters for each band separately. Especially with rough settings, it often makes sense to first adjust all parameters together.

Advanced Mode: When this option is selected, the same internal processing routines are set as in the "Advanced Dynamics" module.

Important: If you have activated "Link bands", it does not mean that all parameters in all bands will have the same values. Only those parameters are the same which were changed after "Link bands" was selected. If all bands should feature the same settings, click "Copy to all bands".

Preview: When this option is selected, the dynamics section works in preview mode. Without preview, artifacts (pumping) and overmodulations can appear. On the other hand, without a preview, this option will not automatically "level" steep attack phases in the waveform, which may result in sharper-sounding audio material. This setting affects all bands.

Bypass dyn. (bypass dynamics):

Dynamic editing of separate bands is removed from the signal route. This function serves to compare the editing results with the original.

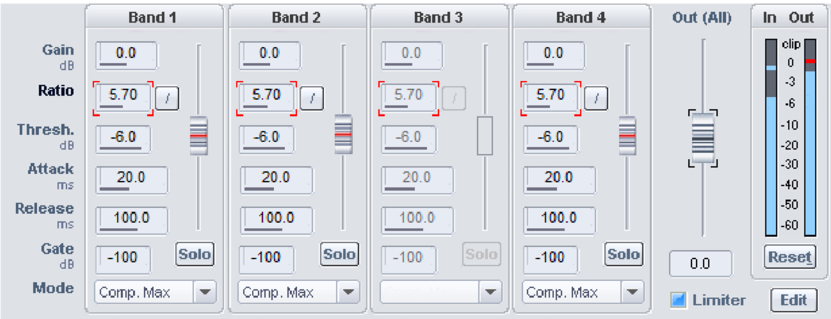
The difference to the "Bypass all" switch is that with "Bypass Dyn." in solo mode, individual bands can be compared separately, with or without dynamic editing as the filter bank is still activated.

Bypass all:

Here, the complete multiband dynamic section is removed form the signal path. Since the entire filter bank is "bridged", Solo mode can no longer be used. The gain knob for the sum is also no longer active.

Multiband Dynamics - set up and parameters

Parameters for the dynamic sections



> **Note:** Depending on the parameter you select by clicking on the corresponding field (gain, ratio, threshold, attack, release or gate), the fader to the left of the input fields controls the corresponding parameters.

Gain (dB): Here you can set the level at the output end of the dynamics phase for each band separately. Please note that the graphic display can cannot be fully display the influence of this control. If you set the controller higher, clipping may occur - In this case the reference line has to be continued upwards out of the display border.

Ratio (factor): This parameter controls the strength of the corresponding effect, 1.0 means that there is no effect. Clicking the button beside it resets the ratio to 1.00.

Threshold (-dB): This is the input threshold value above and below which the corresponding effect is applied.

Attack (ms): Here you can set the time frame between the crossing of the threshold and the maximum extent of the effect.

Release (ms): Here you can set the time frame between the falling off of the threshold and the complete dissipation of the effect.

Gate Level (-dB): This parameter specifies below which volume the level should be set to 0.

Solo mode: In "Solo" mode the frequency bands can be monitored one at a time. This function particularly alleviates adjusting the filter parameters. For instance, you can locate a critical frequency range of a mix before editing the dynamics.

Mode

Comp. Max.: Dynamics of the frequency band is limited in such a way that loud passages above a certain threshold are amplified less, raising the overall volume. These settings are often used to add assertiveness to audio material or to increase the loudness. Among other things, this type of dynamics editing effect is referred to as the "Maximizer" or "Loudness enhancer". Use the "Ratio" input field to set the degree of compression via the "Threshold". You can also influence the "Attack" and "Release" via the corresponding input fields.

Compressor: The effect on dynamics is similar to a classic compressor: The dynamics of a frequency band is limited in that loud passages above a certain threshold are limited in their amplification. Use this settings if you want to achieve sound alteration through compression without raising the overall volume. The level of compression is adjusted by the "Ratio" controller; the activation threshold is determined by "Threshold". "Attack" and "Release" influence the build-up and decay times. A gain control is added to the compressor to compensate for volume reduction.

Expander: The dynamics of a frequency band are increased, loud sequences remain loud, quiet sequences become even quieter. Dynamics expansion is often used for recording speech with a high noise level. The expansion causes the level of the speech to be raised while the noise is suppressed. Please note that there is also a powerful DeNoising algorithm (view page 959) provided.

Gate: Here, very quiet passages below the Threshold Level are dampened or set to zero. This way, you can effectively suppress noise during the pauses between takes. Even at high compression (ratio > 5), the gate function is still useful to avoid strong increases in the quietest of passages and background noise. If different threshold values are entered in different bands, with a bit of skill, drum loops may be "chopped up".

Limiter: Only the loudest passages are limited (above the threshold). Quiet passages remain unchanged. Limiters are used to reduce the occurrence of big level peaks without reducing the master dynamics.

Limiter 100%: Performs the same editing as the Limiter, but the level is immediately raised to 0 dB - this corresponds to a subsequent normalization. This corresponds to a subsequent normalization.

Hint: If you want to use the Limiter as protection against overmodulation, you should take into account that the Limiter can only reliably prevent overmodulation in a single band. If the bands are mixed together, the sum can again result in overstepping the threshold level set for the bands.

Out (All)

Here you can set the overall output of the entire algorithm. This parameter cannot be set separately for each band. The graphic display does not reflect this setting. Use the "Volume" fader to balance the volume difference via the dynamics process.

Limiter On/ Edit:

Here you can add or edit a peak limiter. It prevents overmodulations at the output during volume increases when a multiband dynamics section is used.

Separation frequencies

Click on the graphic display of "**Frequency separation/Level increase/reduction**" in order to change the separation frequencies for individual filter bands by dragging with the mouse. The number of separation frequencies depends on the number of selected bands ("Number of bands" parameter). The border frequencies for the deepest and the highest band (high-cut and low-cut) are shown. The border frequencies are those frequencies where the filter dampening is -3 dB. For mid bands (band pass 1 and band pass 2) the mid frequencies and bandwidths are displayed. The bandwidth in this case is the distance between the two separation frequencies. These also correspond to the cut points of the neighboring frequency curves.

Band separation (in the setup area)

A higher setting for "Band separation" ("Normal", "High") has a similar influence on the various properties of the filter, whereby the "precision" also increases:

1. The edge steepness of the filter curves increases and the transition range between two bands decreases.
2. Dampening in the stop band increases (low: ca. 25-35 dB, normal: ca. 35-45 dB, high ca. 55-75 dB).

3. The ripple of the frequency range of the bands decreases. However, this is not a problem as the ripple of the individual filter bands compensate each other when put together thanks to the complementary filter technology. In any case, the output signal does not contain ripples.
4. The latency caused by the effect increases.

Multiband Dynamics – filter bank for experts

Those who know a little about the program will be curious why the setting/display of values doesn't appear as it does in the parametric equalizer. The answer: in this case, a different digital filter type is used whose frequency response cannot be fully written using these parameters. While analog filters and "normal" digital filters always contain a logarithmic level drop (the drop can, for example, which is set to -12 dB per octave), the filters used in this case have a linear frequency-response curve, i.e. the level drop in dB increases in size the further the distance from the separation frequency. This method also has the advantage of 100% phase accuracy, among other things. The display of the mid frequencies is derived from the "traditional" filter parameters and this way simplifies "acclimatization".

Tips & tricks: strategies to manage the flood of parameters

A multiband dynamics processor (naturally) contains many parameters.

If you need to alter the dynamics of all bands, rather than one individual band, the following approach may be useful:

Step 1: Global presets for all bands

1. Select the mode that corresponds the most with your needs. If you wish to increase the volume of the audio material, select "Compr. max.". If you wish to improve sound characteristics, increase the transparency, improve the clarity of speech, freshen up older recordings or create more powerful bass without increasing the volume, use the "Compressor". The selection of this mode only applies to the selected band.
2. Click "Copy to all bands". Selecting this mode ensures that all bands feature the applied setting.
3. Activate "Link bands". Any changes you perform in the current band are automatically mirrored on the other bands.
4. Change the parameters of all bands until the audio sounds right.

Step 2: Fine-tuning of individual bands

1. Turn off "Link bands".
2. Activate "Solo" mode for a band. Individual bands may now be isolated, making the task of optimizing settings for each band easier. The optimum settings for the higher frequency bands have lower values because the waveforms are shorter here.

3. If you can't find satisfactory settings for some of the bands, try to change the split frequency setting for the selected band. A narrow band setting may help prevent "pumping" artifacts in the sound.

To edit a specific and critical frequency range only, a different approach may be taken:

1. Turn off "Link bands".
2. Activate "Solo" mode for the band that contains the critical frequency range.
3. Change the separation frequencies for the band so that they can effectively filter out the critical area. Now the dynamic editing can begin. First, select a suitable mode.
4. "Limiter" mode or "Compressor" mode are suitable for the dynamic limiting of critical frequencies, i.e. for sibilant sounds.
5. Use the "Bypass dyn." checkbox to compare the processed and unprocessed bands.
6. Now turn off the "Play solo" option and compare the original and edited signals using "Bypass all".

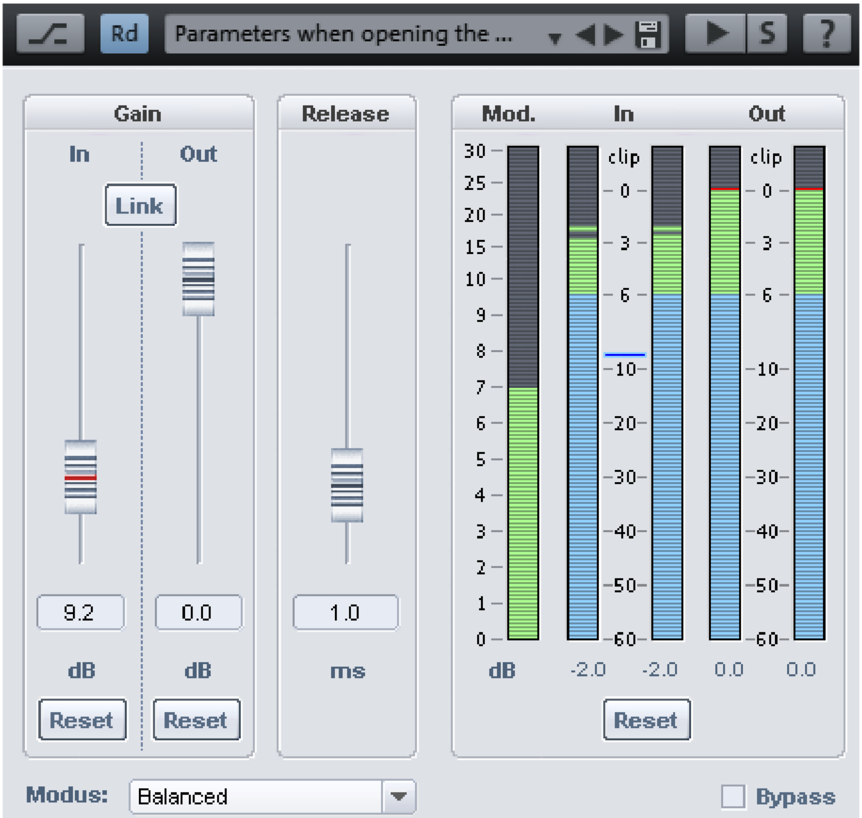
A third option is to first compare the supplied presets and then use the presets as a starting point for searching for optimum settings.

In most cases, even loading a relevant preset will produce good results. But if you really want the very best sound or volume increase result, you'll often have to apply these settings manually.

Comparing at constant volume levels

Use this method to compare the sound differences without allowing the dynamics processing to alter the volume levels. Use the "Out (all)" option to adjust the volume until the audio signal level matches that of the bypassed level, i.e. when the "Bypass all" option is used. If the multiband compressor is used in the "Master" area of the mixer, the peak meter of the mixer may also be used as support.

sMax11



The sMax11 maximizer provides a tool for increasing the loudness of the audio signal.

This is done by entering the input amplitude (**gain-in**). The signal will be amplified by this level. The sMax11 simultaneously ensures that the signal doesn't exceed the output level (**gain-out**) that has been set. This requires regulation of the response time set via the **mode** and **release** time. Essentially, this involves a hard or brickwall limiter with input amplification.

A typical workflow can go something like this:

- First, turn on the **Link** option. This way, volume will stay constant and you can better compare your changes.
- Now, raise the input amplification **Gain in** until the sound changes become unacceptable.
- Adjust **Gain in** to a lower level.
- You can also minimize distortions by raising the **Release** time. However, the compression effect and the volume increase are thereby minimized.

- Switch **Bypass** on once in a while.
- When you've found the optimal setting, switch the **Link** option off.
- Finally, switch the **Output gain** to 0 dB.

sMax11 - General controls and presets

Save, **load** and **delete** functions are integrated into the presets list, where they are available in the lower section. The default file extension is ***.max**.

Play/Stop: This button activates/stops the real-time preview function. This lets you check the acoustics of the respective filter settings.

? (Help): Opens the help file for this effects dialog.

sMax11 - graphical displays

Mod. display: This meter indicates the modification of the signal in dB. The portion of the input amplitude is not included. Only the level with the non-linear changes (required for catching level peaks) will be displayed.

In/out displays: These meters display the input and output displays in dB.

Reset (meter): This button resets the meter displays.

sMax11 - Special controls

Gain-in: This value specifies how much the signal should be increased (maximal).

Reset (Gain-In): Resets the "Gain-in" parameter.

Gain-out: This value sets the maximum output level.

Reset (Gain-Out): Resets the "Gain-out" parameter.

Link: This option connects both gain values.

> **Note**: If the link option is active, then "Gain-in" may be increased, for example, without increasing the output volume. This makes it easier to recognize distortions that appear as a result of a gain-in value set too high.

Release: Set the time frame for the complete dissipation of the effect on the signal in seconds.

Mode: The setting of the mode influences the effect's control behavior: Select the mode most appropriate for your application:

- **Balanced**: Lowest level of distortion with transparent sound, robust to use Appropriate for example for speech recordings.
- **Fast**: Addresses slightly faster than "Balanced". The mode corresponds to the behavior of the hard limiter in the advanced dynamics compressor.
- **Aggressive**: Very short attack time, suitable for especially sharp transients for this reason In signals with dominant percussive components, the sharpness of the beats is preserved. Distortions are masked by the beat.
- **Hard clipper**: Controls are shut off in this mode. Signal peaks are simply cut off, which may lead to significant distortions. This mode is appropriate for example to accentuate the transients of individual beats even stronger.

sMax11 - True Peak Maximizer

Only in Sequoia/Samplitude Suite:

True Peak: The maximum peak level (Maximum True Peak Level) can be selected as a **Clipping Threshold** that is set in the Loudness Normalization. In this way the sMax11 can be expanded to include the True Peak function as defined by the ITU-R BS.1770. Inter-sample peaks are also taken into account during the limiting of the signal so that the reconstruction of the audio material at the analog output does not clip or distort.

efx_Compressor

The **(essential) eFX Compressor (view page 857)** is a simple, efficient tool for reducing dynamics with soft characteristics and adaptive control processing. It is a very musical compressor.

eFX_Gate

The eFX_Gate (view page 859) is a classic "analog" gate, which addresses the signal quickly and accurately while avoiding the typical raw artifacts of digital devices.

eFX_Limiter

The eFX_Limiter (view page 861) is a simple yet effective tool for increasing the loudness of an audio signal. This creates a compressed but loud signal without allowing the defined output volume to be exceeded.

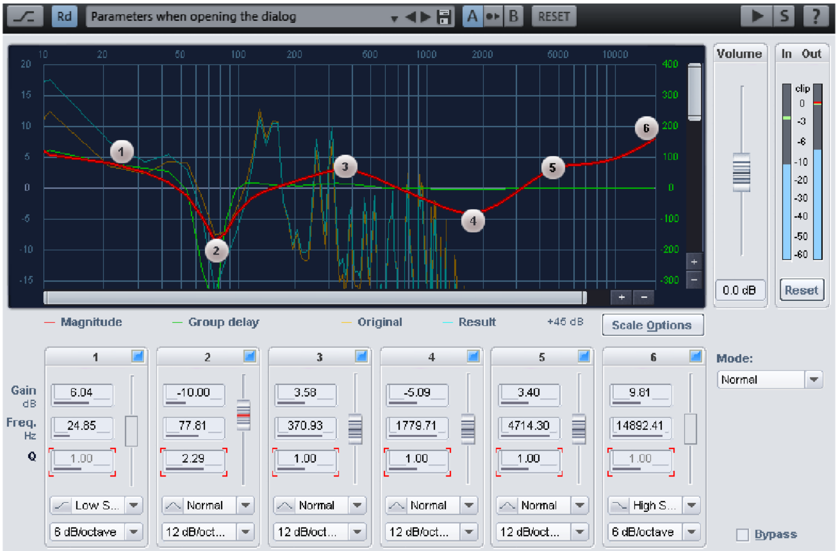
AM-Munition, AM-Track, AM-Phibia, AM-Pulse

(Samplitude Pro X Suite)

More information is available in "MAGIX plug-ins (view page 849)".

Frequency/Filters

EQ116



EQ116 is a 6-channel fully parametric equalizer for audio signal frequency response.

EQ116 - Graphic displays

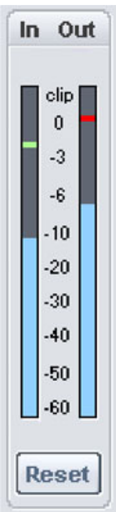
Filter graphic: This graphic provides a view of a number frequency-related details:

- **Amplitude response:** The resulting amplitude envelope from the individual band settings
- **Group runtime/Phase response:** One of these envelopes may be shown optionally. Toggle between these in the scale options dialog. The group delay describes the frequency-dependent time delay in passing the signal whilst the phase response shows the dependence of the phase of the frequency.
- **Original:** This curve displays the original frequency response.
- **Edited:** This curve shows the frequency response after processing by the effect.

The display of the filter graphic may be adjusted by accessing the "**Scale options**" and by using the vertical and horizontal scroll bars.

Hide real-time curves: This option hides the "Original" and "Edited" real-time curves.

Real-time offset: Adjust the offset value in dB for the real-time curves for optimal display. Click in the display window after every change to the values to make the adjustment visible.

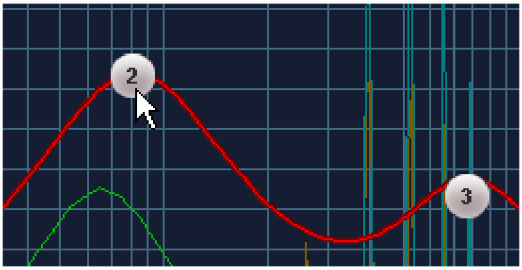


In/Out displays: These meters show the input and output displays in dB.

Reset (meter): Reset the in and out displays.

EQ116 - Editing the reference line

Filter graphic: The band filter parameters gain and frequency may be changed within the graphic by clicking and moving the numbered level points. By turning the mouse wheel you can change the bandwidth (Q) of a band.



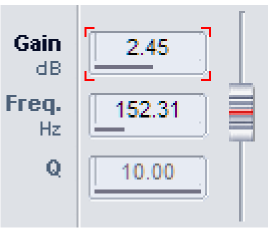
You can change the value of a parameter in the input field by vertical dragging of the mouse or by entering numerical values.

Depending on the parameter you select by clicking in the input field (gain, frequency, or Q factor), the fader to the left of the input fields controls the corresponding parameters.

You can also use the mouse wheel to regulate the selected parameter. If you hold the "Shift" key, the values change in smaller increments.

Volume: Use this controller to adjust the master volume if the level rises or sinks too much while filtering.

Band on/off: In the header for each band group, there is an on/off switch for turning the corresponding band on or off. The associated level point will disappear from the filter graphic accordingly. The corresponding base point will then disappear in the filter graphic.



Gain: These controllers allow you to raise or lower the filter. Setting the controller to 0 deactivates the filter and thereby uses no CPU power.

Freq. (frequency): The frequency controllers set the cut-off frequency of the individual filters between 10 Hz and 24 kHz. You can freely choose the frequency enables multiple filters to be set to the same frequency giving a greater effect.

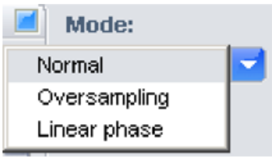
Q (bandwidth): Set the bandwidth of the individual filters between 0.10 (very wide bandwidth) and 10 (extremely narrow bandwidth).

Type: Set the filter type here. These include **normal (= peak)**, **low shelving**, **high shelving**, **high pass**, and **low pass**

Slew rate (gradient): This influences the internal process of a filter band and thereby the slew rate of the high or low pass in the lockout zone. The settings are not equally available to all filter types. For **high pass** and **low pass**, a slew rate may be selected between 6 dB/octave, 12 dB/octave, 24 dB/octave, or 36 dB/octave. For **low shelving** and **high shelving**, select between 6 dB/octave and 12 dB/octave. The filter type "**Normal (= peak)**" is fixed at 12 dB/octave.

EQ116 - Internal operation modes

Mode: Set the internal working mode of the EQ116 here.



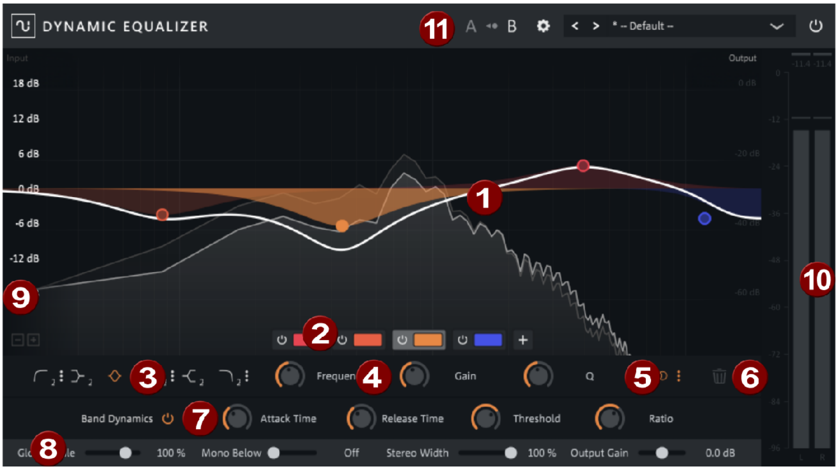
The following modes are supported:

- **Normal:** Equalizer functions in normal operating mode, which corresponds to a minimum phase EQ. The mode corresponds with the parametric 4-band equalizer found in older versions of Samplitude, and therefore represents a compatibility mode.
- **Oversampling**: This effect works internally compared to the first mode and with a higher rate where the filter formation takes place. Especially for higher frequencies, a more precise frequency response can be reached, and fewer aliasing artifacts arise. The disadvantage is that this mode requires more computing power.
- **Linear phase:** Equalizer works in such a way as to eliminate frequency-dependent phase displacements. This process is completely different than in the first two modes. The linear phase mode sounds more neutral and less colorful, and the influence on the amplitude of the overall signal is weaker because no amplification and extinguishing effects appear as a result of phase offset. However, it does command higher computing power. In addition, for impulse-heavy material with high frequencies a special artefact, called "pre-ringing", can take place.

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 (view page 769) 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.

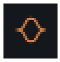


1. **Filter curves and filter symbols:** Each filter band is symbolized by a colored circle. The filter curve of the individual band is represented by the transparent colored area below the circle. The thick white line shows the resulting filter curve of all bands. In the background, the light gray line shows the resulting frequency spectrum of the signal in real time, and, somewhat paler, the original spectrum of the signal.

2. **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.

3 **Filter Type:** Here you set the filter type of the selected band.

 High pass  Low shelving  Peaking  Band pass

 High shelving  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).

4 **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.

5 **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

> Note: This feature is not available in surround busses/masters.

6 **Delete filter band**

7 **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**

1 **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.

2 **Settings:** Here you can disable the tooltips.

3 **Previous/next preset**

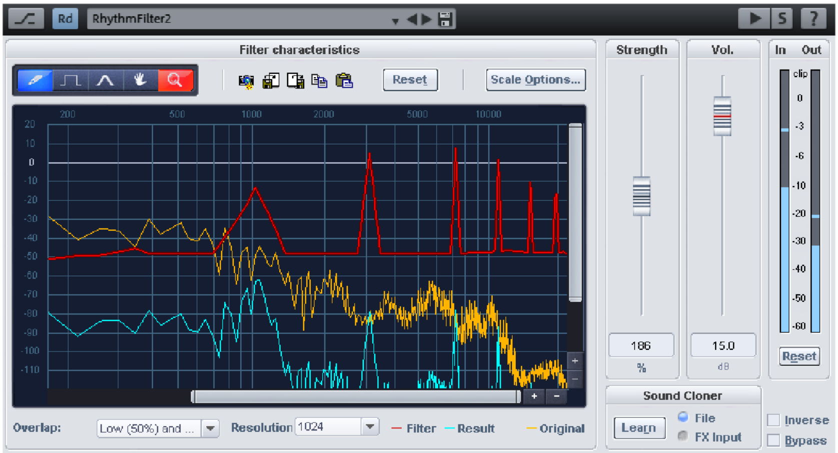
4 **Preset menu:** In the menu you can select the presets from different categories.

5 **Bypass:** Use this button to temporarily disable the effect.

Tips:

- If you move the mouse over the filter parameters, the value is displayed numerically. By double clicking on this value, it can also be entered numerically. Use the Tab key to move to the next value.
- New bands can also be created directly in the filter graphic by double-clicking. Depending on the location of the click, the correct filter type is selected: Left above 0db -> Low Shelving, Left below 0dB -> Highpass, Right above 0dB -> High Shelving, Right below 0dB -> Lowpass, Center -> Peaking.
- Right-clicking on a circle for a filter band brings up a context menu that allows you to cut, copy and delete filter bands. A copied or cut filter band can replace another one with **Replace**, with **Replace (using current position)** the frequency and gain remain unchanged.

FFT Filter



The FFT filter (for real time use in the object editor, in the track or master section) allows you to precisely filter linear signals.

Caution: When using the FFT filter as an object effect, the latency of the project increases by several milliseconds per object due to the linear-phase mode of operation. If you are working on long projects and have used the FFT filter on a very large number of objects on a single track, this latency can add up to several seconds. This means it can take several seconds before audio is actually played back after playback start.

Edit the red filter curves by drawing with the pencil tools in the graphical display. If you let the object play along, you will immediately hear how the sound of the edited object.

The graphical view shows 3 curves that are integrated in a coordinates system. This includes the vertical axis, which shows the volume in dB, and the horizontal axis, which shows the frequency in Hertz (the pitch).

The **yellow curve** represents the **original frequency response**, and the **blue one** represents the **corrected frequency band** that accompanies the original frequency response after the **red filter curve** is applied.

Hint: The filter has a bigger drawing area when used as an offline effect ((Menu Effects > Process effects offline).

The following tools are available via the object FFT filter:

Pen for free-hand drawing: This tool allows you to draw the filter curve. Hold the "Shift" key down to draw straight lines.

Pen for quantized drawing: Use this tool to draw stepped filter curves; the quantization steps depend upon the set resolution. Hold the "Shift" key down to draw straight lines.

Bend tool for curving lines: Click with this tool above or below the curve to bend the curve/frequency area. The further away from the line you click with the bend tool, the sharper the curve will be. This effect can be intensified by simultaneously holding down the "Ctrl" key.

Navigation tool: Use the navigation tool to move the visible selection vertically and horizontally while zoomed in.

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

Besides the FFT filter tools, the following buttons are also available:

Snapshot: If this feature is activated, the filter's current input and output spectrum will be preserved in the graphical interface.

Save: This command saves the filter curve as a text file.

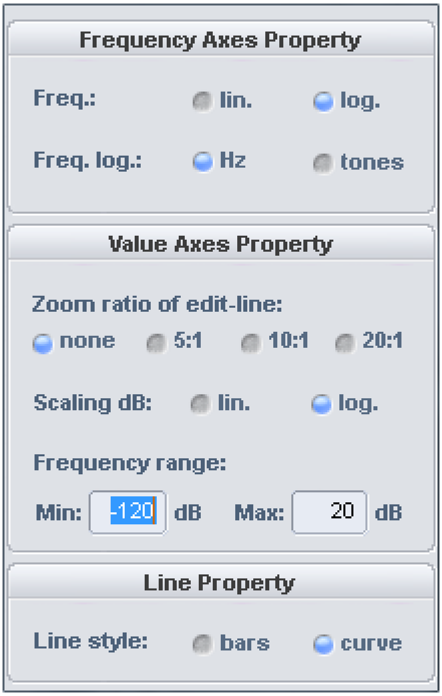
Load: This command loads a filter curve from a text file.

Copy: This command copies the filter curve to the clipboard (keyboard shortcut: Ctrl + C).

Insert: Use this command to insert the filter curve from the clipboard into the chosen FFT filter of the selected object; this replaces the current filter curve in the graphical interface.

Reset: This button resets all curves for the FFT Filter.

Scaling options: Set the graphical display of the curve value here.



Frequency axes property: The frequency can be output as a linear curve or as a logarithmic curve. If the logarithmic display is selected, then the display's x-axis can be output either according to frequency or according to pitch.

Zoom ration of edit line: Here you can stretch the filter curve on the y-axis, either 5:1, 10:1 or 20:1. For the dB values choose either the linear or the logarithmic display. The range of values can be defined freely. Simply enter the desired minimum and maximum values into the corresponding fields.

Line style: Display the curve roughly as a stepped bar or as a soft curve.

Level: This controller compresses or stretches the filter curve. Set the level of adjustment for the transmitted frequency response between 1 and 200%.

Vol.: This controller adjusts the total volume of the filter curve in dB, and thereby the "Out" value of the level display.

FFT filter/Sound Cloner functions

With the Sound Cloner feature you can determine the sound characteristic of a selected object and transfer it to another one. For instance, if individual songs of a music collection are different to others, the sound cloner feature can adapt them to other pieces. An 80's hit could be treated with the characteristic sound of the late 60s, for example.

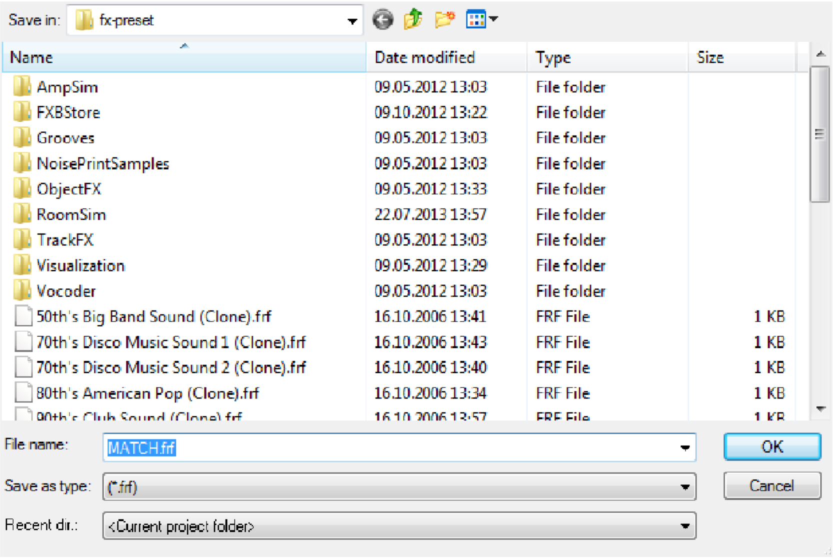
First of all, determine the spectral image of the object that should be used as an audio template. Save this as a "clone". The clone may now be applied to the target audio material.

The sound cloner's preset menu also contains a few useful filter settings besides the "clone" presets.

Step 1: The learning process

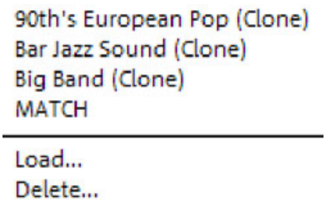
Load a song with a sound you would like to use it as a reference. Place the play cursor at a location in the song where the elements of the sound to be cloned appear. Don't select the intro or a break. All of the instruments involved should be playing at this point. Press the "Learn" button.

The options "File" or "FX input" specify whether the sound clone should originate from the original wave file or from the object that has been processed with effects. The sound characteristic of the reference song is now created as a fingerprint. Save the sound clone that has been created in the "FX presets" folder.



Step 2: Transfer frequency response

Now select the frequency response of the object that should be adjusted and open the FFT filter. By loading a **match preset from the preset menu**, the sound characteristic will be communicated to the sound at the position of the play cursor.



The filter curve is calculated via the target audio material's spectral analysis, which provides an audio image that is similar to the audio image in the preset. If used as an object effect, the target material to be analyzed is taken from the associated object, in which case the play cursor specifies the position inside the object. The "Level" controller regulates the intensity of the sound transmission.

When loading the sound clone, it's important to place the play cursor at a location in the object where all of the instruments are playing; the sound cloner uses the audio material at the play cursor as the basis for calculation of the filter settings. The selected filter curve will not be transferred directly into the FFT filter; initially, a spectral analysis of the target audio material is executed at the play cursor before the selected object's filter curve is adjusted. By default, 20 seconds will be analyzed, but the length of the area to be analyzed can be adjusted by dragging with the mouse on the range in the arranger.

FFT filter/Sound cloner controls

Learn: Calculates a frequency analysis at the playback position. A dialog opens in which the result can be saved as a special match preset.

File: The sound clone's reference is created using the selected object. This produces a spectral image of the object. Next, the Sound Cloner preset can be saved and then appears at the bottom of the presets list. Real-time effects are ignored in this mode when defining the reference.

FX input: The input at the effect is used as a reference. Realtime effects looped before the FFT filter which is used as a cloner are accounted for in this mode when defining the reference.

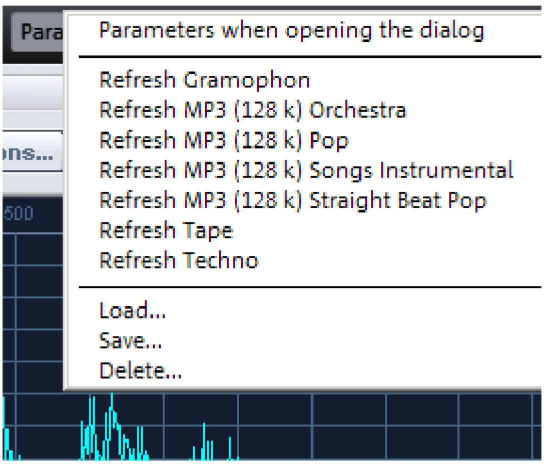
Inverse: Inverts the filter curve. This function inverts the sample data within the selected range along the amplitude axis, i. e. negative values become positive and vice-versa.

Bypass: The algorithm is removed from the signal route. In this way, the unedited signal can be compared to the result of the algorithm.

Brilliance Enhancer

(Optional as part of the Cleaning/Restoration Suite and Samplitude Pro X6 Suite)

Brilliance Enhancer – Presets



Save, **load** and **delete** functions are integrated into the presets list, where they are available in the lower section. The default file extension is ***.bre**.

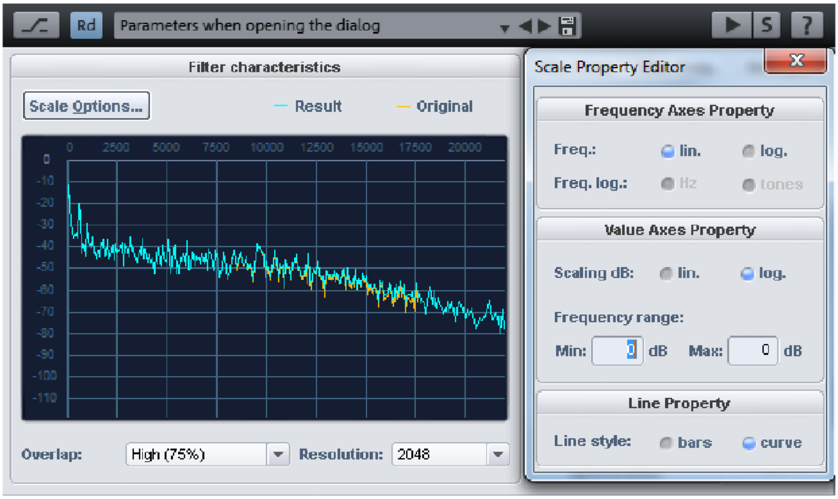
Brilliance Enhancer – fine settings

You will see two curves in the graphic display:

- The orange curve is the original signal
- The blue curve is the edited signal

Scaling options: This button opens the "Scaling options" dialog, which can be used to adjust the signal display to your needs.

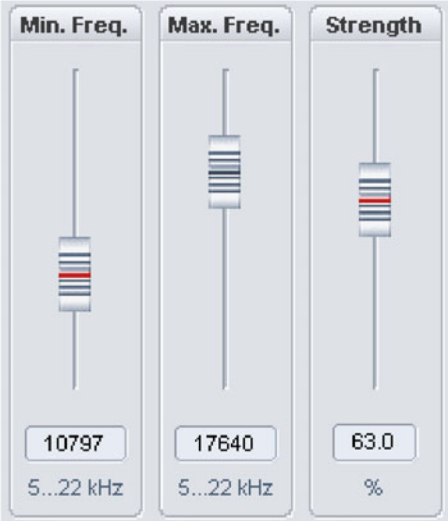
Value/curve display: With the help of input and output spectrum you can decide how the scaling options have an effect on the display during playback or monitoring.



Overlapping: This internal parameter changes the overlapping of the time window for the modulator signal spectrum calculation. Higher values improve the result, but also increase CPU load.

Rate: This parameter sets the internal rate used by the algorithm. You may wish to experiment with this parameter because a higher resolution does not always guarantee the best results.

Brilliance Enhancer – parameters and controls

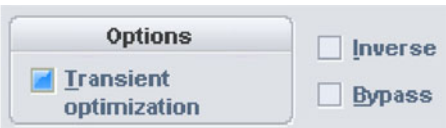


Minimum frequency: Here you can set the lower limit for the frequency band which will be enriched with new overtones.

Maximum frequency: Here you can set the upper limit for the frequency band which will be enriched with new overtones.

Intensity: This parameter is used to set the intensity, with which the new overtones should be introduced into the audio material.

Options



Transient optimization: If this option is activated, transients will be freshened up by the addition of higher frequencies. Older or compressed pop and jazz recordings benefit especially from this setting. The setting is less appropriate for tonal material, as odd overtones may result.

Inverse: In this button is activated, you will hear only the newly-generated part, the overtones being added.

Bypass: The algorithm is removed from the signal route. In this way, the unedited signal can be compared to the result of the algorithm.

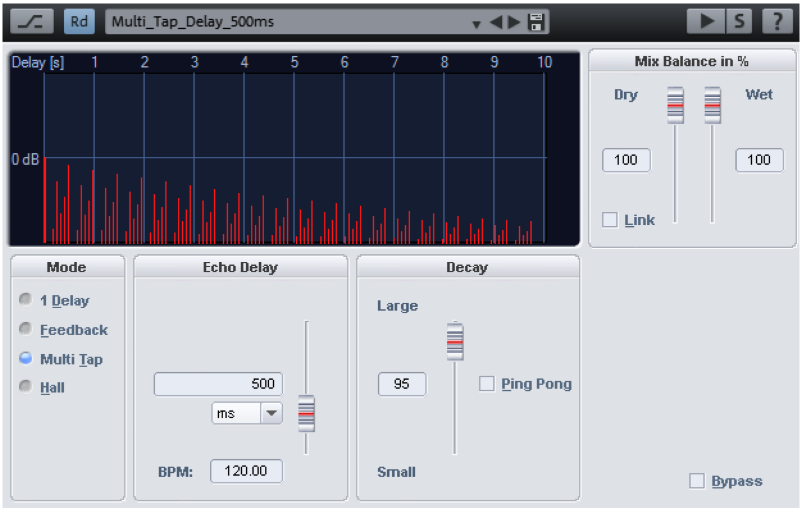
Filtox

More information is available in "Effects -> MAGIX plug-ins (view page 849)".

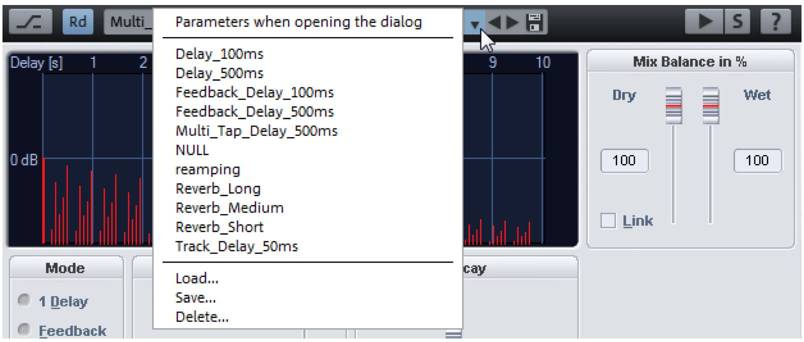
Delay/Reverb

Delay

With this effect, you can embed an echo or reverb into an audio file or object.



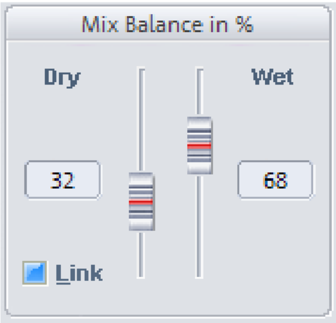
Delay presets



Load, **save**, **delete** presets: Here you can save, load, or delete settings. The default file extension is ***.ech**.

Delay parameters

Mix Balance in %: Here you can enter the ratio of the original signal to the echo/reverb signal in percent. If you switch on the "Link" option, both values always add up to 100%.

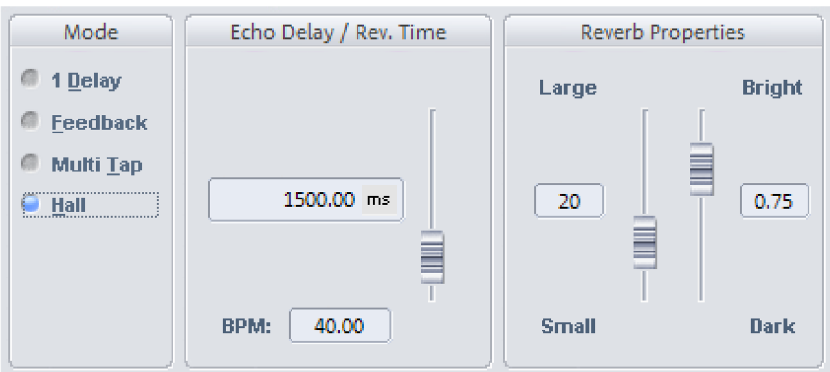


Mode: With mode you can select the basic character of the effect. The following modes are available:

- Individual delay
- Delay with feedback
- Multi tap delay
- Reverb

Echo Delay/Reverb Time: This fader sets the delay between single echoes or the original signal and the first echo in milliseconds. The fader controls the reverb time in "Reverb" mode.

BPM: Enter the current song beat in BPM, which automatically sets the delay value that corresponds to a quarter note in this tempo.



Decay/Reverb Tonality: Here you can set the decay value for the simulation of the size of the room or reverb tonality numerically as well as using faders. A ping-pong effect is also available for the delay modes. In "Reverb" mode you can set the reverb room between "Small" and "Large" and "Light" and "Dark" using two faders.

Bypass: The algorithm is removed from the signal route.

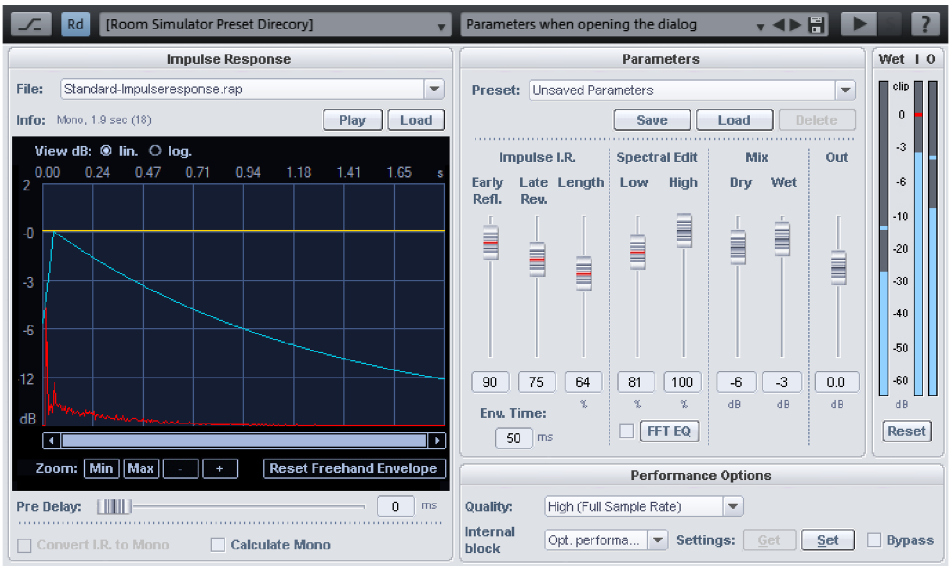
Room Simulator

This function allows you to simulate the reverb qualities of various rooms. The impulse response or reverb time of a room is calculated with a audio material.

A room impulse response is the reverb of a explosive noise, for example like that of a pistol. It contains all the necessary information to simulate room reverb accurately.

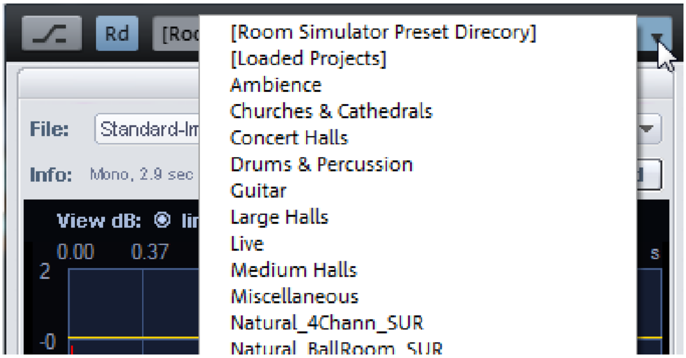
You can manipulate the impulse response using the dialog parameters. In this way you can adjust the reverb properties in a similar fashion to how digital reverb-effect devices do this. In addition, however you also have the ability to set the reverb character by fundamentally defining the impulse response.

A graphic representation of the impulse response and the envelope allows an optical overview of the impulse response manipulation using the Room Simulator parameters dialog. In Samplitude this effect is also available as a realtime effect in object, track, AUX bus, and master channel.



Room simulator – banks and presets

The room simulator offers a range of reverb banks in categories such as "Medium halls" and "Miscellaneous", in which various presets and impulse responses are saved.

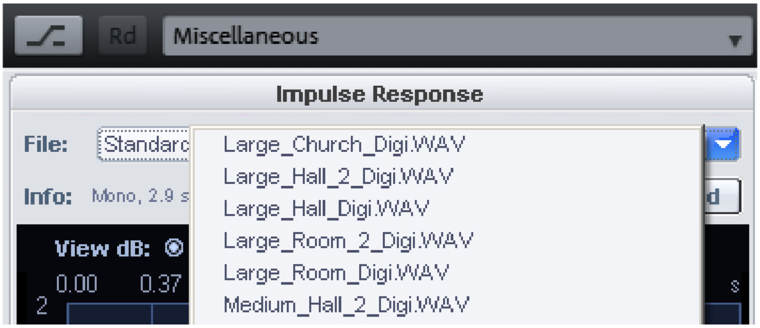


When a preset is loaded, the parameter settings of the dialog field and an impulse response are loaded. The impulse response can be any sample, which is saved on the hard drive as a wav, hdp or rap file.

Included impulse responses have ***.IMR** extensions.

The presets have the extension ***.rms**. Files with the impulse responses of the included presets are found in the "fx-preset > RoomSim" folder.

Presets found in the "RoomSim" folder are organized according to their affiliation with a certain bank in the preset list and are loaded by selecting one of the list entries.



Clicking on "Load" lets you retrieve preset files from other folders or from the room simulator.

Impulse response - controls

File: Select an impulse response. Impulse responses from the "RoomSim" folder and the audio files opened in the program (bank: [Loaded projects]) are available.

> **Note:** If during Samplitude installation you have deactivated the "Install impulse response" option, you will have access only to the opened audio files (Bank: [loaded projects]) as well at the "Standard-impulse-response.rap" from the "Room Simulator preset directory" bank.

Info: Here you can get information about the length of the impulse response and about whether it will play back in mono or stereo.

Play (Impulse response)

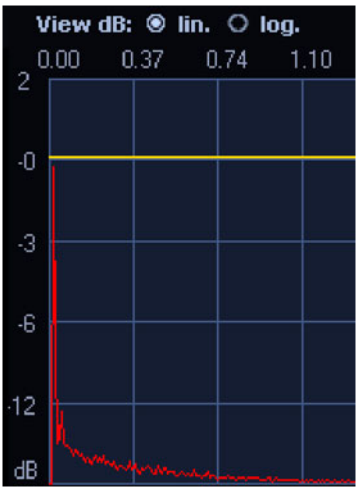
The impulse response will play after pressing this button. Impulse responses from IMR files will be played back only with 16 bits, even if the impulse response has a 32-bit float format.

Load (Impulse response)

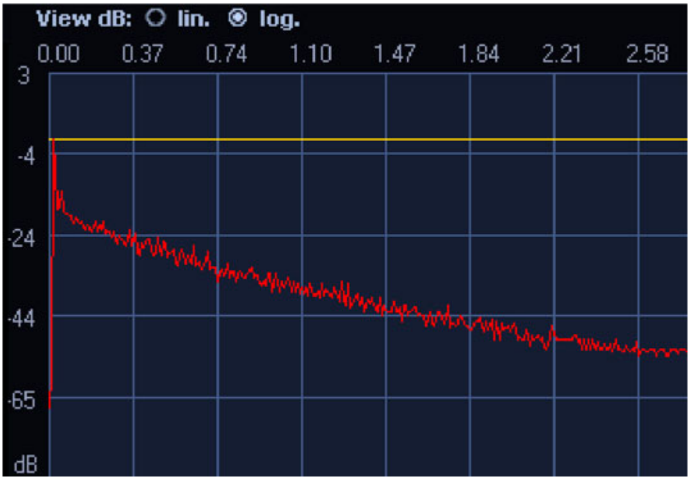
The impulse response can be any sample that is saved to the hard disk as a WAV, HDP, IMR or RAP file. from any folder on the hard disk

Graphic display

dB lin display: The graphic display with the impulse response has linear amplitude scaling. This settings corresponds to the common display of samples.

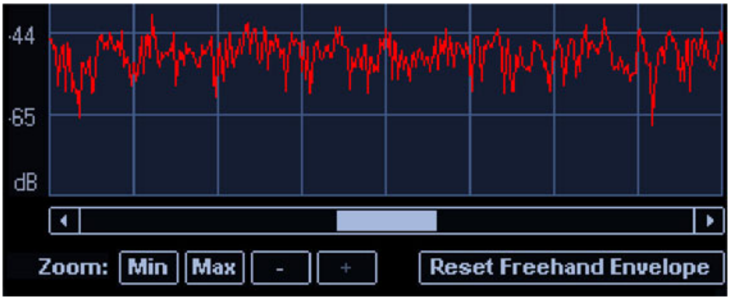


dB log display: Choosing this setting will scale the amplitude display of the impulse response logarithmically. This corresponds to the way the human ear perceives volume changes.

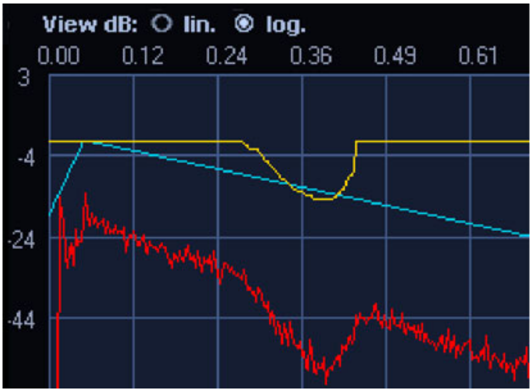


Zoom options (All, In, Out, Max, Position):

This option lets you zoom into the display and select the range to be displayed.



Curves



Red: The red curve represents the impulse response.

Light blue: The light blue display is the two-segment envelope curve to dampen reverb. The front segment is the earlier reflections, and the later segment is the post-reverb.

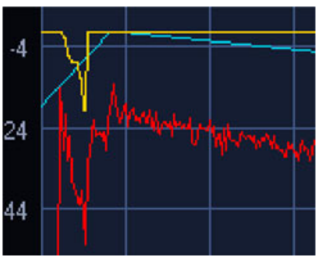
Yellow: The yellow curve is a freehand drawable envelope curve.

Freely definable envelope curve

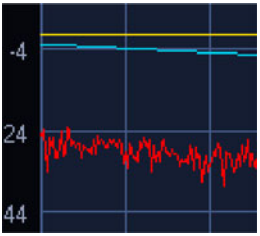
The impulse response scale can be manipulated with a yellow freehand envelope curve. Typically, it is used to dampen or delete early reverberations of the impulse response.

To edit it, click on the yellow curve with the mouse cursor and drag it with the mouse key held down.

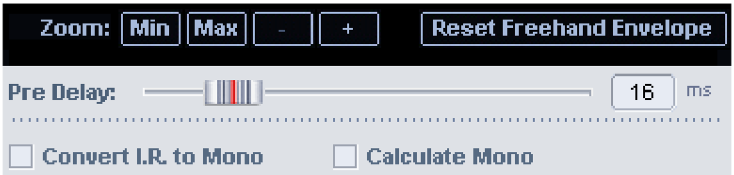
The use of zoom options makes it possible to work precisely.



By clicking on the "**Reset freehand curve**" you can reset the yellow envelope curve.



Pre-delay



The "Pre-delay" function delays the entire impulse response by a time interval variable between 1 and 100 ms.

Convert impulse response into mono

Selecting "Convert I.R. in mono" recalculates stereo impulse responses into mono. Convolution is calculated on two channels as before, in Surround Mode corresponding to the count of the group channels.

Calculate mono

Convolution is only calculated for one channel which reduces the required processing power. The input signal and the impulse response (if stereo) are converted to a mono signal before convolution. For surround application, all channels of the group are added to one mono signal.

Room Simulator - parameters

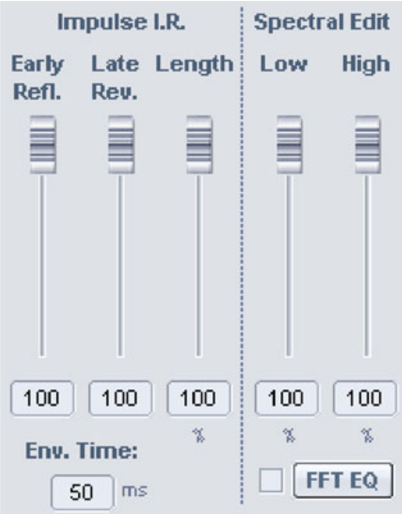
Parameter presets

Select, save, load, and delete presets that consist of parameters for envelope curves, frequency sequences, and the mix here.

The presets are useful for testing various parameter settings without having to change the impulse response.

The parameter presets have the extension *.rsp.

Envelope, EQ



Early reflect: This parameter allows you to lower or suppress early reflections by dampening the first part of the impulse response.

Late Reverb: This parameter lets you decrease or suppress the final part of the impulse response.

Length: This setting can shorten the length of the reverb effect to up to 5% of the original length by shortening the impulse response pattern. Please keep in mind that the reverb can end rather abrupt, which may lead to unnatural decay patterns. You can compensate by applying a lower Late Reverb setting to fade the impulse response with the 2 segment envelope curve.

Env. Time: Here you can set the time length of the first of the two segments of the envelope curve. Using this parameter, you can influence the damping of the earlier reflections.

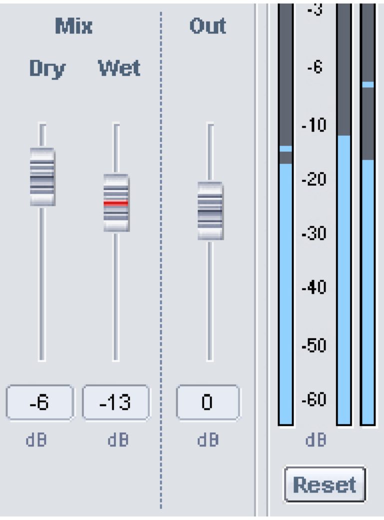
Spectral Edit: Parameters for editing reverb frequencies

Low: Using this parameter you can adjust the low frequency component of the reverb.

High: Using this parameter, you can adjust the high frequency component of the reverb.

FFT EQ: The reverb component can be edited with an additional **FFT Filter** (view page 788). The graphic display with realtime spectrum of the reverb component in the FFT Filter allows optical control of the frequency characteristics of the room, where the impulse response was recorded. Undesired resonancies can be quickly opened and removed.

Mix, Volume, Reset



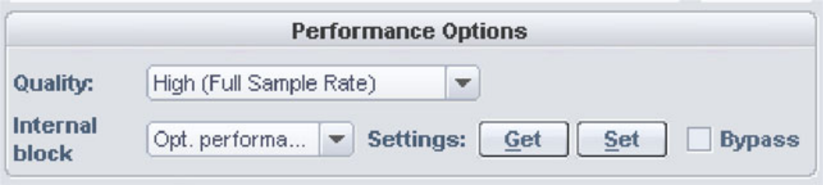
Dry: Here you can set the original signal level in dB.

Wet: Here you can set the reverb level in dB.

Out: Here you can set the output level in dB.

Reset: Here you can reset the held highest levels on the level display.

Room Simulator - Performance Options



Quality: In both "**Normal**" modes **"Normal"** and **"Normal Plus"** room simulation is calculated only with half of the sampling rate. In most cases this will be quite enough, since natural impulse responses and even impulse responses generated by digital reverb devices rarely posses components above 10 kHz (you can check this in the spectral representation of the integrated FFT Filter). (You can check this in the spectral display of the integrated FFT filter). Many older reverb devices work internally with half sampling rate, making calculation of the entire frequency range a waste of CPU capacities.

The difference between the two "**normal**" modes lies only in the quality of resampling that leads to sampling rate reduction. In "Normal plus" mode, the resampling quality is higher. The required calculation performance is slightly raised in this mode.

In the **"high"** mode, the entire frequency scale will be calculated. The CPU load doubles in comparison to the **"normal"** mode.

Internal Block Length: This parameter specifies a block length with which the convolution function can be calculated internally.

Short block lengths raise the count of the required calculation operations, causing CPU strain to increase. Long blocks lead to irregular CPU loads. The parameter has no effect on the calculation results.

The optimum for realtime processing tends to be about **4096** to **32768** samples.

Note: For use in the AUX, track or master without low latency, the setting of 2048 or lower is recommended on faster systems for an acceptable response latency.

For use in low-latency conditions, the value can be reduced to **32** on faster systems.

The value "**optimal latency**" adjusts the internal block length in such a way as to keep the latency as low as possible without influencing the performance too much.

The value "**optimal latency**" adjusts the internal block length in such a way as to keep the performance as high as possible without raising the latency too much.

Hint: It doesn't make sense to set a value smaller than the set ASIO buffer size. If the value corresponds to the ASIO buffer size, the room simulator processing is latency-free (only with "higher quality").

When using the room simulator for offline processing, the block length parameter is raised internally, as processing with a lower latency would in this case result in a useless increase of the necessary CPU operations.

Retrieve, Set: Opens and sets the quality options globally. When you have found a fitting setting, click on "Set". This saves the setting. Now you can continue to experiment. If you want to revert to the saved settings, click on the "Retrieve" button. This restores saved quality options.

Bypass: If you activated "Bypass" you will hear the unprocessed signal.

Room simulation - Tips and tricks

By using the numerous functions for destructive sample/object editing in Samplitude, you can alter the qualities of the impulse response in countless ways by using the Room simulator dialog.

- If you reverse apply the "Effects > Sample manipulation > Reverse" function to the impulse response, you will get a reverse reverb effect.
- By using the Timestretching function the size of the room can be altered without affecting the resonance behavior of the room itself.
- Employ an explosive envelope on any short samples and use the result as an impulse response - a great way of creating exotic reverb effects
- You can also get interesting reverbs by using the impact on a percussion instrument as an impulse response.

Room simulator - problems and solutions

Reverb sounds unnaturally hard.

You can control the impulse response by setting the "Reverb" parameter. The graphic display of the impulse response makes fast optical control possible especially in logarithmic display (dB log).

The signal created contains a strong DC offset.

Remove DC offset with the same function; "Remove DC offset (view page 835)"

Heavy hard disk activity or an error message concerning insufficient memory was received.

The algorithm needs lots of storage space, especially with long impulse response patterns. All background processes should be terminated and all projects that are not needed should be closed.

CPU power usage is too high.

- Check the "Internal block length" parameter setting.
- Set the "Quality" option to "Normal".
- Shorten the impulse response length.

Problems with very long impulse responses.

In realtime application, the impulse response length is limited in principle only by CPU capacity. For offline calculations, the limit is 380.4 seconds (at 44.1 kHz sample rate). If this upper limit is exceeded, the samples behind it are ignored on calculation. Impulse responses of this length are, however, not practical for room simulation. If you would like to experiment around with very long impulse responses (longer than 1 minute), you should note that the algorithm usually requires a lot more RAM. When loading long audio files as an impulse response, a warning will appear noting that calculation will take a lot of time.

eFX_Reverb

Detailed information on eFX_Reverb can be found at "essential FX > Reverb (view page 853)".

eFX_StereoDelay

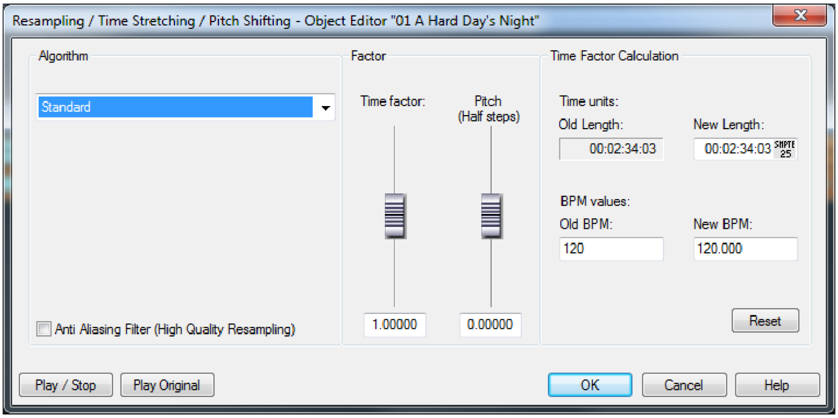
Detailed information about StereoDelay is available in "essentialFX > StereoDelay (view page 855)".

Ecox/VariVerb Pro

More information is available in "Effects > MAGIX Plug-ins (view page 849)".

Time/Pitch

Resampling / Time Stretching / Pitch Shifting



All algorithms in this dialog use the "time factor" and "pitch (semitones)" as input parameters.

Open the "Time factor calculation" section to determine the required time factor from the original length and original tempo as well as the desired new length/desired new tempo.

Tip: To change an audio file's sample rate, e.g. from 48 kHz to 44.1 kHz, use the function "Effects -> Sample manipulation -> Adjust sample rate (view page 848)".

Resample

Samplers and PCM Synthesizers transpose samples using this procedure. Time factor and pitch are dependent upon each other: the shorter the audio material, the higher the pitch and vice versa. The effect is comparable with changing the playback speed on record players or tape recorders.

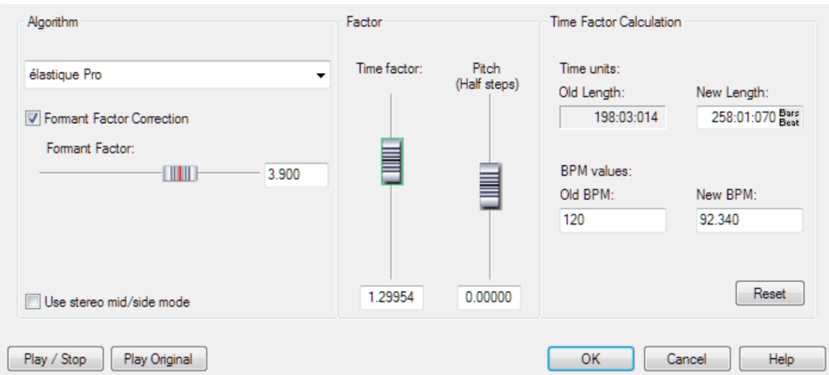
Resample is a relatively lossless process. If a pitch change resulting from the change in speed is wanted or acceptable, you can use this algorithm.

Timestretching/Pitchshifting

With all other algorithms, time factor and pitch variation are independently adjustable. These algorithms create, under circumstances, artifacts which you can balance using the Anti-aliasing feature. The preset timestretching/pitchshifting algorithm can be saved via Timestretch/Pitchshift Patcher in Wave project (view page 719) ("Object" menu).

élastique Pro

With this algorithm licensed from zPlane you can achieve impressive results with minimal additional information such as "Time Factor" and "Pitch". As with the „Monophonic Voice“ (view page 810) algorithm, this mode can also be used to preserve or shape formants.



Use stereo M/S mode: This option enables M/S processing. This means that the algorithm internally interprets the adjacent stereo signal as a mid signal in the left channel and a side signal in the right channel.

élastique Efficient

When real-time application of élastique Pro threatens to overload the CPU this algorithm offers a less CPU-intensive alternative. However, if you choose this option, be prepared to make some compromises regarding sound quality.

Monophonic voice

This is a special Time Stretching and Pitch Shifting function for vocal solos, speech, or solo instruments. The audio material should not contain background noise – strong reverb may also this method's effectiveness. With suitable audio material, the audio quality will remain high.

The "**Use formant correction**" option preserves formants during Pitch Shifting, i.e. the "Mickey Mouse effect" does not occur. This allows realistic-sounding background choirs to be compiled using one solo voice. The formants, however, may also be moved by +/- 12 semitones in order to achieve desired vocal distortions.

Typical uses of this algorithm include:

- Intonation correction: The note with the imprecise pitch should be cut out as an object so that it may be manipulated independent of the other notes.
- Harmonizer effects: Copy the object containing the singing. If the pitch is altered now, a second voice will be generated.
- Creating background vocals from existing vocal samples
- Time Stretch distortion of a spoken sample, e.g. grandfather's voice.

The "Monophonic voice" algorithm is especially useful for: Speech, single-voice singing, single instruments without overlaps/with low levels of reverb and background noises.

Elastic Audio (Real-time)

Elastic Audio – general

Elastic Audio is a specialized editor for changing the pitch of audio material. It uses a combination of automatic resampling and pitch-shifting algorithms as well as pitch detection for monophonic material.

Elastic Audio offers the following options:

- Detection of the basic frequency in monophonic audio material.
- Automation of resampling and Pitch Shifting Algorithms. This also includes an algorithm for format-true pitch shifts of monophonic audio material that utilizes results of the pitch detection basic frequency analysis.
- Automatic and manual correction of the basic frequency process in monophonic audio material.

- Automatic and manual correction of the pitch of notes from monophonic audio material.
- Changing melodies in monophonic audio material.
- Manual correction of the basic pitch curve.

Note concerning Time Stretching: Time Stretching cannot be automated, but it can be used statically with Elastic Audio.

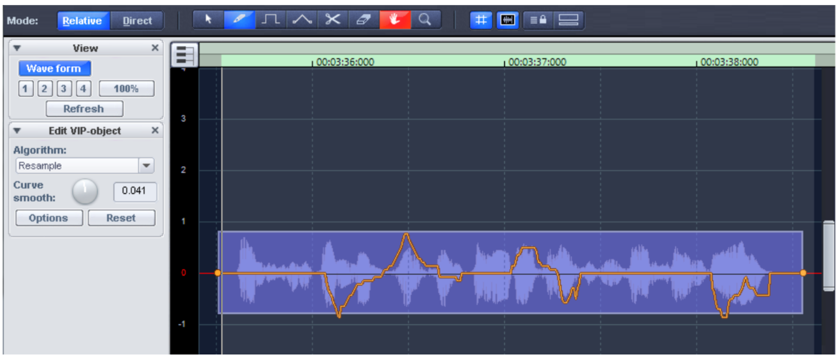
Elastic Audio - editing modes

The Elastic Audio editor may be used in two different operational modes, i.e. **Relative** and **Direct** mode. If Elastic Audio should always open in "Direct" mode, then select the function **Open in mode "direct"** from the Elastic Audio "View" menu.

At the lower part of this menu you can see which controls are available and switch them on and off.

Elastic Audio – "Relative" mode

Edit the pitch curve (automation of the pitch's temporal path) as a relative pitch deviation here. This corresponds roughly with the pitch bend controller for MIDI data.



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

Overview of graphics in "Relative" mode

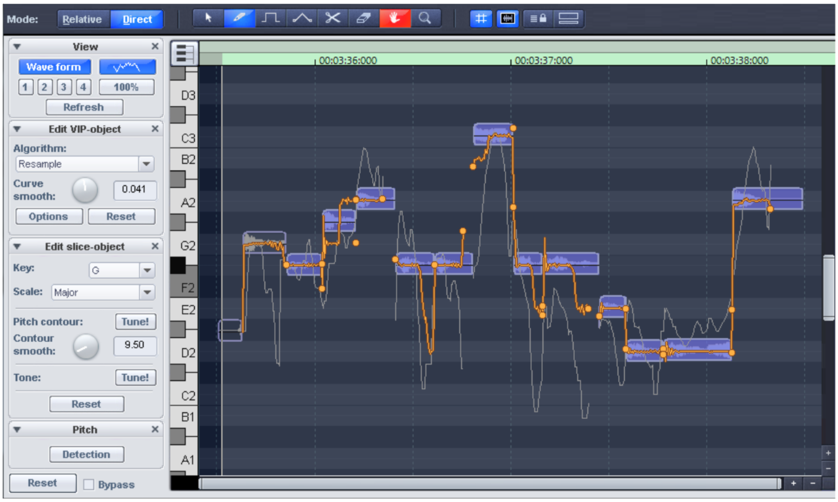
Y-axis: Display of pitch changes in semitone steps (within a range of -24 to 24 semitones).

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

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

Elastic Audio – "Direct" mode

The pitch curve is traced immediately in "Direct" mode, and changes are absolute. To customize the pitch correspondingly, the original pitch of the audio material must be known. A preceding analysis of the audio material's pitch is therefore required for "Direct" mode. This is basically only for tonal, monophonic material like solo vocals, solo instruments, and speech.



Start the analysis function with the "Pitch Detection" button. The analysis may take longer with larger objects. After analysis, the VIP objects are divided into individual slice objects according to the recognized pitches. The medium pitches of a slice object determine its position in the graphic, independent of the set progression of the pitch curve inside the slice object.

Select all of the slices first. Click an empty area in Elastic Audio Editor to deselect all of the slices. The individual or multiple slices may now be selected via the modifier keys "Ctrl" and "Shift".

Two handles are created for the pitch curve on the borders of the slice objects. These handles can be moved in order to produce an increasing or decreasing pitch characteristic, but still keeping the small changes in the basic frequency (vibrato).

Besides the pitch trace feature, this mode also provides you with "Tune!" functions for automatic pitch correction.

Overview of graphics in "Direct" mode

Y-axis: The piano keys on the left side of the window display the pitch assignment in notes. The individual pitches may be deselected by clicking them in this view so that they won't be used in the automatic pitch correction or in quantized drawing.

You can get appropriate scale settings by selecting the tonic keynote and the scale in the “Edit slice-object” box.

Orange line: Displays the editable pitch properties.

Gray line: Displays the original pitch properties.

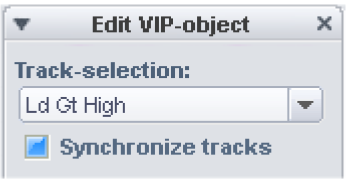
Blue line: Median pitch of a slice object.

Apply Elastic Audio to objects on multiple tracks.

To do this, select the objects you wish to edit in Elastic Audio in the Arranger. The objects can come from different tracks.

The Elastic Audio Editor loads all objects that were previously selected. When objects are loaded into Elastic Audio Editor from several tracks, switch the track displayed in the editor with the option "Edit VIP objects" in the track selection.

With the "Synchronize tracks" function in the Relative mode, you can apply pitch changes of the active object in the track selection to all other loaded objects, even if they are found on different tracks.



Example:

Imagine if you distributed choir recordings over multiple tracks.

Step 1: Select all choir recordings to be edited on various tracks in the Arranger.

Step 2: If you now open Elastic Audio, you can use track selection in Relative mode to pick which recording from which track is displayed, and edit it first.

Step 3: Activate the "Synchronize tracks" option, so that all pitch changes also effect other selected choir recordings on other channels.

Elastic Audio: Control Elements

In the lower section of the "View" menu of Elastic Audio Editor, you can see which control elements are available in each mode and switch them on or off.

View

Waveform: This turns the waveform display on and off.

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

Save zoom levels:

Shortcut:	Ctrl + Numeric keypad 4
	Ctrl + Numeric keypad 5
	Ctrl + Numeric keypad 6

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

Get zoom level:

Keyboard shortcut:	Numeric keypad 4
	Numeric keypad 5
	Numeric keypad 6

Left-clicking on the zoom button while holding down the Shift key overwrites the last zoom level.

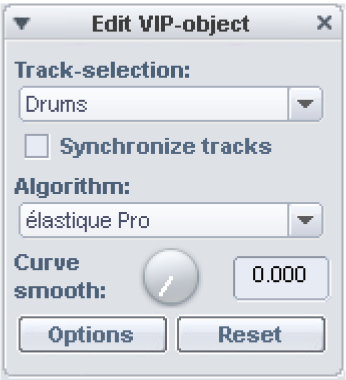
Left-clicking on the button while holding down the Ctrl key opens only the horizontal position of the respective zoom level.

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

Note: The key combination "Ctrl + Alt" + Mouse wheel provides simultaneous horizontal and vertical zooming.

Refresh: This updates the graphical display of the material. Resulting changes to length are adapted. When actively synchronizing the arranger and Elastic Audio Editor (Elastic Audio Editor "View -> Horizontal -> Synchronize VIP") the effects on the VIP are also displayed in the arranger window.

Editing a VIP Object



Track selection: If objects in multiple tracks have been loaded into Elastic Audio Editor, they can be selected here.

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

Detailed information about these algorithms is available in "Effects -> Time /Pitch -> Resampling/Time stretching/Pitch shifting (view page 808)".

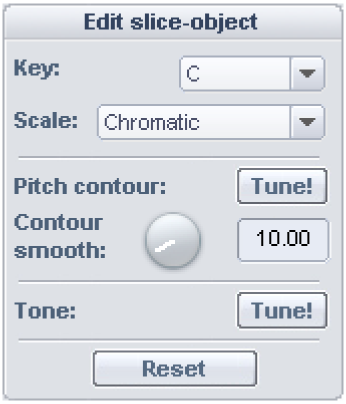
Curve smoothing: This parameter brings about the smoothing of the given pitch curve using a time constant (in ms). When smoothing to a large extent, even jump-like pitch shifts can are changed to a "glissando".

Options: These are advanced settings that depend on the selected algorithms.

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

Options for automatic pitch correction

Edit slice object (direct mode)



Key: This option sets the basic key of the scale. In the chromatic key, this setting is not included in calculation.

Scale: Use this option to select the type of scale, i.e. the sound make-up/mode. Choose from "major", "minor", "harmonic minor", "pentatonic", and "chromatic".

Pitch contour: Tune!: This button quantizes ("levels out") the pitch contour of the selected slices.

Contour smooth: Set the strength of the quantization (lower values quantize hard). This function removes repeated small pitch alternations in natural sound sources, e.g. vibrato ("Cher" effect).

Tone: Quantize the mid pitch of the selected slices. The corresponding sections of the pitch contour of a slice object are moved together and adapted to the mid pitch. Pitch fluctuations within a slice object remain the same.

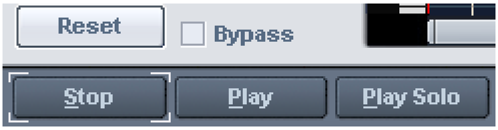
> **Note**: Automatic correction requires the slice's pitch (pitch -> detection).
>
> The results of the calculation of mid pitches for slices featuring glissandos often do not correspond to the pitches that human hearing assigns to them. In this case, automatic correction of the mid pitches does not give satisfactory results. Removal of the glissando passage by manually changing the slice borders (keyboard shortcut: Ctrl + rubber handle tool) or the splitting of slices can help.

Reset: This command resets the selected slices. The orange curve is superimposed over the gray curve and the slice is reset to the original mid pitch.

Pitch (fundamental frequency)

Detection: Use this button to start the basic frequency analysis. In this case, the absolute pitches are displayed graphically in the Elastic Audio Editor. This function is indispensable for additional editing steps you may wish to make in "Direct" mode. For this reason, you should only open the pitch recognition before executing pitch corrections or other editing processes.

Playback control



Reset: All pitch curves for the entire material loaded into the Elastic Audio Editor are reset.

Bypass: Plays the arrangement without editing.

Stop/play: Stops/starts the playback of the arrangement.

Play solo: Only activated objects in the editor are played.

Toolbar

Several tools are available for processing the slices and pitch envelope. Two tools can each be assigned to one of the two mouse buttons. The tool assigned to the left mouse button turns blue; the function assigned to the right key is marked in red. Click with the corresponding mouse button on the desired console button. The "Zoom" tool is an exception: both buttons are assigned automatically.

Selection tool (arrow): Use this tool to move slice objects vertically. This modifies the pitch of slice objects as a whole. Slice objects and curve handles may also be selected, and a slice object lasso can be dragged out. Multiple selections are possible with the "Ctrl" or "Shift" key.

Freehand drawing function: Use the pencil tool to draw the pitch curve freely. If the "Shift" key is pressed, a straight line is drawn from the starting position to the current position of the mouse. When the "Ctrl" key is also pressed, the slice objects are combined during drawing.

Pencil tool for quantized drawing: This is the tool for quantized drawing. Quantizing means that the line is free horizontally, but snaps into semitone steps vertically in "Relative" mode, and to the tones of the scale selected via "tune" in "Direct" mode.

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

Rubber handle tool: This rubber band tool may be used to bend the pitch envelope between two neighboring handles. The range between the curve handles is moved, but the handles themselves remain unchanged. The curve is bent inwards at the handle point. In combination with moving the curve handles on the slice object's boundaries, you can bend the tone envelope and thereby maintain the microtonal structures (vibrato).

Cut: Use this tool to manually split the audio material into slice objects. After automatic pitch recognition has been applied, use the "Cut" tool to split certain passages into individual slices. To recompile slice objects, use the pencil tool to drag along the slice while holding down the "Ctrl" key.

Eraser: The orange curves are set back to their initial values, i.e. the pitch envelope corresponds to its original curve again.

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

Zoom tool: Click with the left mouse button to zoom into the display; zoom out with the right mouse button. Left clicking and dragging stretches the range displayed as the zoom area.

Controls for global display options

These buttons have a global effect for the Elastic Audio Editor.

Grid: Use this function to switch the raster on or off.

"Single object" mode: The maximum horizontal zoom level is limited to one object.

"Synchronization" mode for VIP and Editor: Horizontal zoom levels of both windows are synchronized with each other.

"Docked view" mode: When "Docked view" is active, the arranger window is located below the Elastic Audio Editor.

Using Elastic Audio Editor

Elastic Audio: Overview of keyboard commands and configuration of the mouse wheel.

Navigating with the mouse wheel

Horizontal scrolling	Mouse wheel
Vertical scrolling	Ctrl + Shift + mouse wheel
Horizontal zoom	"Ctrl" + mouse wheel
Vertical zoom	Alt + mouse wheel
Zoom horizontally and vertically	Ctrl + Alt + mouse wheel

Keyboard shortcuts

Ctrl + space bar	Play solo/stop
Space bar	Play/stop
Ctrl + A	Select all
A	Refresh view
Ctrl + Z	Undo
Ctrl + 1 - 8	Select tool for the left mouse button
Ctrl + Shift + 1-8	Select tools for the left mouse button.
Shift + Alt + P	Show/hide pitch curve
Ctrl + Alt + T	Show/hide other loaded tracks
Ctrl + cursor up	Direct mode
Ctrl + cursor down	Relative mode
Cursor left	Move play cursor to left
Cursor right	Move play cursor to right
Ctrl + Cursor left	Horizontal zoom in
Ctrl + Cursor right	Horizontal zoom out
Shift + R	Change scale tuning (Frequency for “A” concert pitch).
Shift + A	Show/hide "View"
Shift + O	Show/hide "Edit VIP object"
Shift + F	Show/hide "Pitch"

Shift + S	Show/hide "Edit slice object"
Ctrl + number pad 4, 5, 6	Save zoom snapshot 1, 2, 3
Alt + number pad 4, 5, 6	Load zoom snapshot 1, 2, 3 without vertical zoom
Number pad 4, 5, 6	Load zoom snapshot 1, 2, 3 without vertical zoom

Tone pitch automation including changes to the length (resampling)

- Load objects into "Elastic Audio Editor -> Relative mode".
- Select the "Resampling" algorithm.
- Select the drawing tool.
- Draw the orange automation curve with the drawing tool.

Pitch automation without change in length

Proceed here as in the previously explained resampling section. You can also select a pitch algorithm. Here the length of the VIP objects remains the same.

Pitch correction (intonation correction) with monophonic audio material

- Load the object into the Elastic Audio Editor. Mode -> "Direct"
- Perform pitch detection ("Detection" button in "Direct" mode).
- Select desired slices
- Select suitable algorithm
- Edit pitch with the mouse tools

Manual correction of the whole pitch of a pitch slice

- Select the slice with the selection tool (arrow).
- Move the orange line vertically with the mouse tool.

Automatic correction of the whole pitch of a melody slice.

- Select the slices with the selection tool (arrow). Mode -> "Direct"
- Detect pitch
- Select the scale in the "Edit slice object" group and, if necessary, discard the additional tones that are not to be quantized via the keyboard on the left side of the editor by clicking and deselecting.
- Click the Tone "Tune!" button

Automatic correction of the pitch characteristic with quantization

- Select the slices with the selection tool (arrow). Mode -> "Direct"
- Detect pitch

- Select the scale in the "Edit slice object" group and, if necessary, discard the additional tones that are not to be quantized via the keyboard on the left side of the editor by clicking and deselecting.
- Click the Tone: "Tune!" button
- Reduce level of the quantization with the parameter "Contour smooth"

Correct pitch increase or decrease

- Select the slices with the selection tool (arrow). Mode -> "Direct"
- Detect pitch
- Move the orange-colored pitch curve to the edge of the slice object by moving the handles or click and drag the orange line

Create frequency modulations like warblers and vibrato

- Select the slices with the selection tool (arrow). Mode -> "Direct"
- Detect pitch
- Select the drawing tool
- Draw pitch modulation

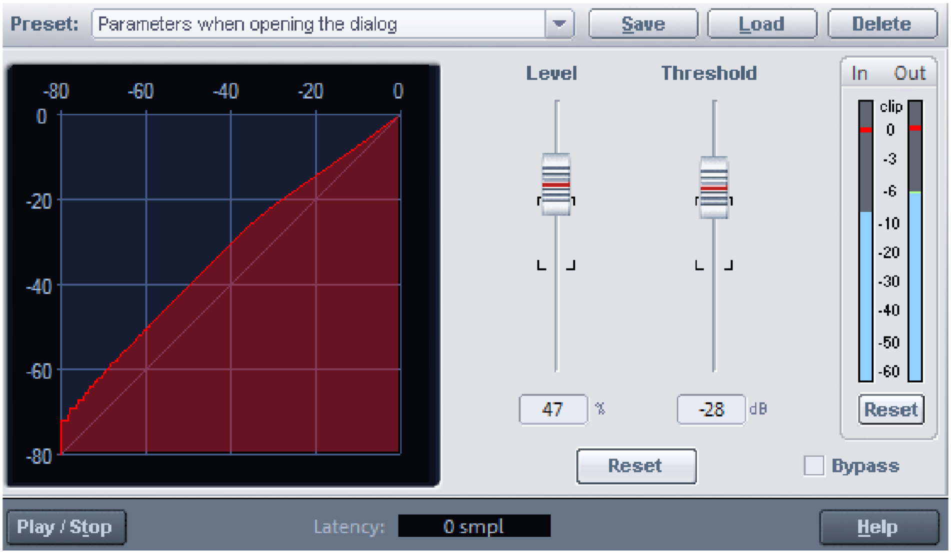
Creating second voices or harmonizing effect

- Create a duplicate of the object on the track below it in the arranger
- Selection of both objects in the VIP
- Open Elastic Audio Editor
- Select the track via "Edit VIP object -> Select track"
- Edit each track individually. Do not use the resampling algorithm to avoid different tempos in the voices.

Alternatively, additional voices may be created easily with the "Harmonizer" in "Direct" mode.

Distortion

This effect allows you to distort audio material by using a non-linear transfer reference line. The signal becomes louder and overtones are added.



By adjusting the starting point of the distortion (threshold), a soft, analog-sounding distortion (overdrive), or a hard, digital-sounding distortion (threshold at 0 dB). The intensity of the distortion may be set using "Level".

Vandal

Detailed information about Vandal is available in "Vandal (view page 919)".

BitMachine

Audio material can always be edited into high quality with Samplitude. Nevertheless, there are some situations, for example, a more imperfect lo-fi sound would perfectly suit a drum loop or a synthesizer sound.

Remember, for example, the first hardware samplers from the 80s that usually only ran at 8 or 12-bit rates and at low sample rates. With the BitMachine, changing the sound with such an “antique” device is no problem.

You can use the BitMachine to bring back to life the times when minimalist and scratchy soundchips in home computers were commonplace.

The BitMachine opens up a gateway to “acoustic time travel” where you can encounter bit and sample rate reduction and downstream filters based on analog models.

Furthermore, the effect has a modulation section with which you can control individual parameters using an oscillator (LFO) or the input signal.

We have designed a range of “typical” presets to demonstrate the time travel abilities of the BitMachine. These can be opened at the top right of the interface

The following section describes the details of BitMachine:

"Reduction" section

Bits

This dial controls the resolution of the audio material. Turning the dial to the left results in 16-bit quantization (CD quality). The further it is turned to the right, the lesser the signal dynamic becomes. In extreme cases (1-bit), there are only “on” or “off” states.

At the intermediate levels, you’ll notice an increase in the background noise and a decrease in the dynamics. For example, 8-bit quantization will exhibit dynamics of only 48 dB. Quieter points in the material sound noisy and very quiet points sound “capped.” This effect is amplified the more you turn the dial to the left until it starts crackling or “groaning.”

Sample rate

The audio material is “down-calculated” with this dial, i.e. the internal sample rate is reduced. A new separation ratio between old and new rates is created. In relation to this ratio, a sample from the data stream will be “dropped” at the various points.

Note: The two smaller dials from this section are explained under **Modulation**.

"Filter" section

The filter in the BitMachine is a digital model of one of the most well-known filters in music electronics, i.e. the “Chamberlin 2-pole” filter used in old Oberheim synthesizers. These types of filters sound exceptionally musical. They can also be used quite creatively in the BitMachine, but should not be used exclusively to smooth out existing artifacts.

The filter works in the so-called “high-pass” mode, i.e. it lets through deep frequency (or medium) material according to setting, and dampens highs and medium areas.

Freq:

You can specify the cut-off frequency of the filter using “Freq.” Filtering starts above this frequency.

Reso:

The signal in the area around the cut-off frequency can be strongly elevated to just below self-oscillation. Sharp, cutting sounds are possible at this level, and the effect becomes even clearer when you vary the cut-off frequency.

Drive:

Both of the individual filters of the connections mentioned above have the ability to overmodulate themselves internally. With the “Drive” dial, you can regulate the amount of overmodulation. The more you turn this dial up, the more the signal is overmodulated. In this case, the parameters of the internal workings of the filter interact with one another. Increasing drive weakens the resonance, but, at the same time, the signal gets more volume, more bass and becomes acoustically fuller.

Note: The two smaller dials from this section are explained under “Modulation.”

"Modulation" section

You can automate your effects via the settings in the modulation section.

Here, you’ll find the so-called low frequency oscillator (LFO), which resonates with adjustable speed. You can influence the speed and type of resonance.

To influence the resonance, use the two small dials in both the reduction and filter areas. These four dials display modulation targets.

Example: You've left the dial for the sample rate at its default setting. Change the small dial beneath from its middle position to either side. The modulation for the dial value is added to the sample rate: The LFO now controls these parameters proportionately and the sample rate reduction resonates at this modulation.

You can use this technique on other dials as well. You just have to make sure that the main dial isn't turned up to full, because then the modulation wouldn't have any effect. The modulation is always added to the set value.

Example: Turn the small dial beneath the "bits" dial fully to the left (Value: -50) and the one beside it (beneath "sample rate") to the right (+50). You've now assigned a modulation to both parameters with the LFO. They are not changed uniformly, but rather opposite to one another: A negative setting is nothing more than an inversion of the modulation, so you're effectively turning down the control signal.

Waveforms of the modulation section

We've already explained this example with the help of sine oscillation. The LFO can be in:

- Sine form
- Square wave (0 or 1, no intermediate level)
- Random value (an internal randomizer will be queried at the set speed)

Oscillator speed

The LFO speed is specified with the "speed" dial. If the "sync" button is active, then the LFO adapts to the song speed, and the dial locks musical values into place (e.g. ¼ note). Rhythmic paths of the sound distortion are therefore enabled. You can also switch off this synchronization and set the speed manually (in Hz).

Modulation with the "Envelope follower"

In the modulation section you'll find a fourth button, the audio input signal. If this mode is active, then the signal itself can be called upon to extract "modulation tension"; a so-called "envelope follower" continuously scans the volume of the input signal.

> **Note:** The BitMachine doesn't recognize the type of audio signal automatically. For this reason, you should set the input sensitivity roughly with the "gain" dial. To do this, use the control LED: With accurate detection of the signal dynamics, assigning the four small dials to modulation lows is easier and you can use the full control range.

In envelope mode, the “speed” dial is used to control the response speed of the envelope (the display now switches to milliseconds). Lower times result in a faster response, higher times make the envelope rise (and fall) slower. You should experiment with the signal according to its complexity. The presets provided can only point you in a rough direction.

eFX_Tube Stage

Detailed information on eFX_Tube Stage can be found at "essential FX > Tube Stage (view page 862)".

Restoration

In Samplitude the following restoration tools are available as standard:

- DeClipper SE
- DeClicker/DeCrackler SE
- DeHisser SE
- DeNoiser SE

You can also remove noise from an object with the mouse mode "Spectral mode (view page 111)". Editing can be done directly in the Arranger window.

Note: With the optional "Cleaning/Restoration Suite" (see below (view page 951)) you have access to the full versions with an enhanced range of functions.

- Declipper
- DeClicker/DeCrackler
- Dehisser
- DeNoiser with Noise Print Assistant
- Brilliance Enhancer
- Spectral Cleaning

SE dialogs header

In the presets input field you can load, save or delete presets.

On the right next to the Preset selection you will find the Bypass and Automation buttons.

Bypass: The algorithm is removed from the signal route. This enables the unedited signal to be easily compared to the result of the algorithm.

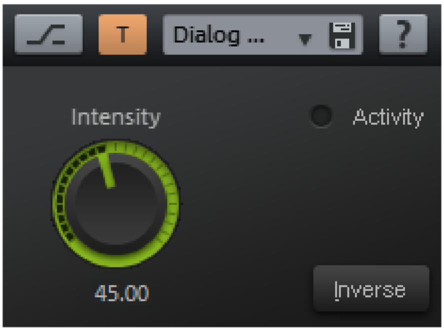
By clicking on this button you can toggle between the selected automation write/read mode. By right clicking you can get to the Automation context menu.

Via "?" you can call up extra help information.

DeClicker/DeCrackler SE

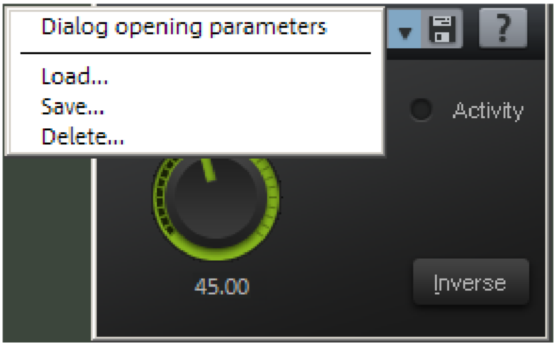
(The full version is available as an option of part of the Cleaning/Restoration Suite)

The DeClicker removes crackling and clicking noises, which are typical on scratched records, it was also developed especially for louder individual clicking sounds on older vinyls.



DeClicker/DeCrackler SE - Presets

Load, **save**, **delete** presets: Here you can save, load, or delete settings. The default file extension is ***.dck**.



DeClicker/DeCrackler SE - Parameter

Intensity: Using this parameter you can influence the intensity with which the DeClicker/DeCrackler SE engages with the detected areas of the audio material.

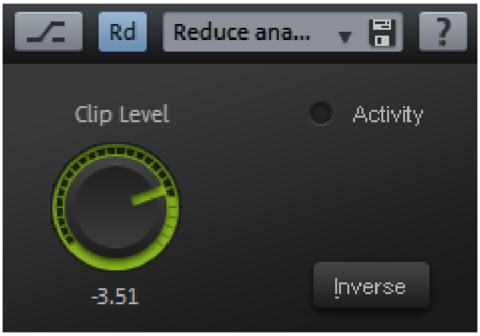
Activity: This LED flashes red if the signal is altered using the DeClicker SE. The LED flashes red if the signal is altered using the DeCrackler SE.

Inverse: If this switch is activated, you will hear only the part of the signal that is removed by the algorithm. If the parameter is optimally tuned you will only hear the bothersome clicks. If the parameters are too high, parts of the music or speech signal will also be filtered, which will lead to music discoloration.

DeClipper SE

(The full version is available as an option of part of the Cleaning/Restoration Suite)

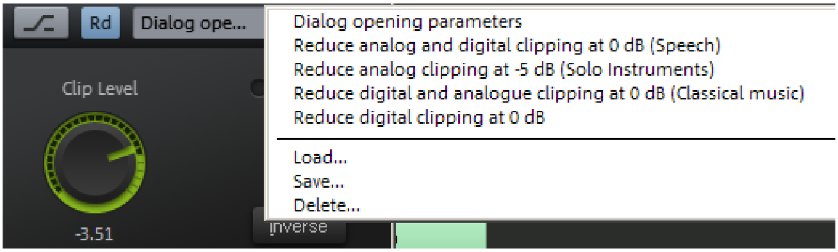
The De-clipper is a tool for removing overmodulation and distortions. Overmodulated passages are recalculated, or interpolated, based on the material immediately surrounding them.



The DeClipping algorithm is suitable for material with clearly audible overmodulation, e. g. distorted piano or vocals.

DeClipper SE – Presets

Load, **save**, **delete** presets: Here you can save, load, or delete settings. The default file extension is ***.dcp**.



DeClipper SE – Parameter

Clip level: Here you can enter the level at which the sample will be considered overmodulated, and correspondingly corrected.

Activity: This LED flashes blue if the signal is altered using the effect.

Inverse: If this switch is activated, you will hear only the part of the signal that is changed by the algorithm.

DeHisser SE

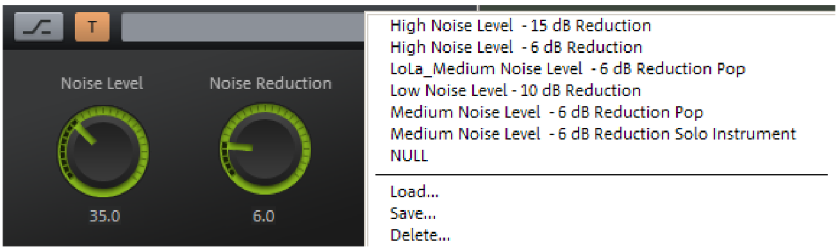
(The full version is available as an option of part of the Cleaning/Restoration Suite)

This command opens the DeHisser SE, which helps to remove noise. The DeHisser SE eliminates regular, low-level "white" noise, typically produced by recordings, microphones, pre-amplifiers or transformers.

Unlike with noise reduction, a sample of the distortion is not required.

DeHisser SE – Presets

Load, **save**, **delete** presets: Here you can save, load, or delete settings. The default file extension is ***.deh**.



DeHisser SE - Parameters

Noise level: 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.

Noise reduction: Set the noise damping here. The highest possible damping value is 30 dB.

In practice, extremely quiet passages, like the slow fading out of one instrument prove to be critical – here the noise level can exceed that of the audio signal.

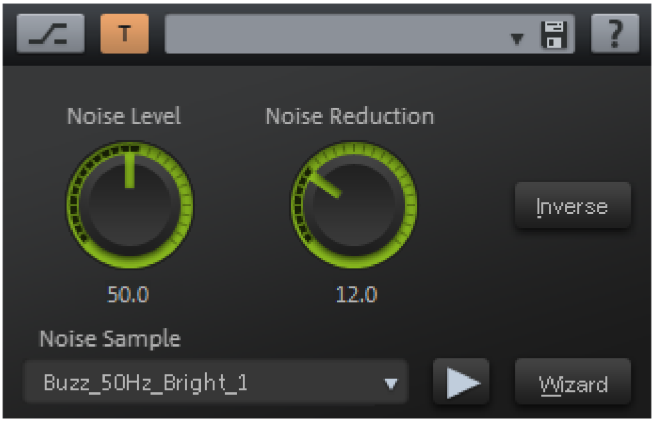
Limit yourself to one low noise damping (-10, -15 dB) to avoid side effects like artifacts.

Inverse: If this switch is activated, you will hear only the part of the signal that is removed by the algorithm. By optimally setting the "noise level" parameters, you should only hear the noise.

DeNoiser SE

(The full version is available as an option of part of the Cleaning/Restoration Suite)

With the DeNoiser SE you can remove noises without discoloring the source material too much. To do this, the algorithm requires a noise sample.



The DeNoiser SE function works especially well to remove even long-lasting noise like ventilation noise, artifacts from low-quality sound cards, tape deck noise or feedback.

DeNoiser SE – Presets

Load, **save**, **delete** presets: Here you can save, load, or delete settings. The default file extension is ***.nr2**.

DeNoiser SE – Parameter

Noise Level

This parameter brings about a increase/decrease in the noise sample spectrum. Be sure that a higher-level of noise does not automatically require a higher correction level.

Noise Reduction

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. Removing background noise from on location reports is usually not preferable. Besides, any artifacts or discolorations are reduced by the incomplete suppression of noises. Think of the process in terms of reduction, not 100% elimination.

Inverse: If this switch is activated, you will hear only the part of the signal that is removed by the algorithm. If the parameter is optimally tuned you will only hear all the noise and only a small portion of the used signal

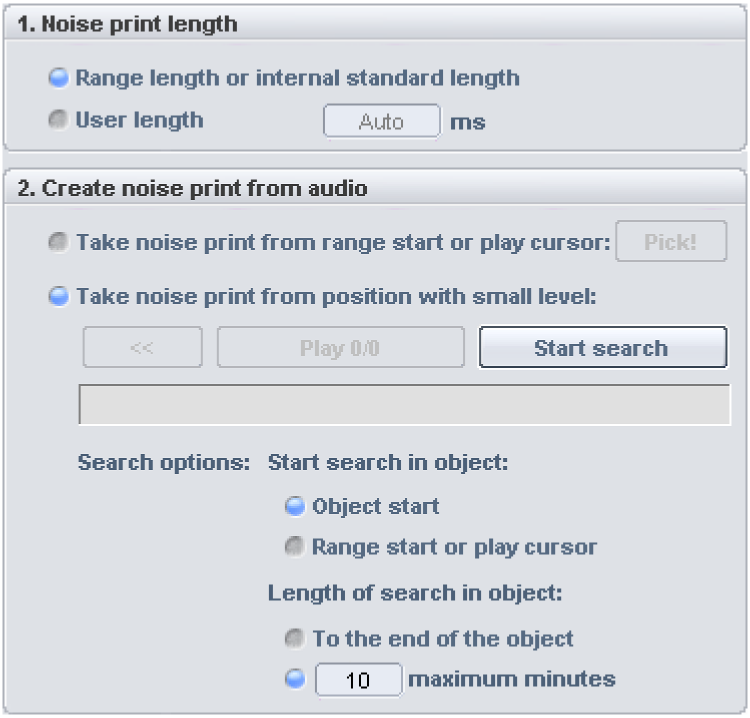
Noise sample file: In the selection list, you can select a group of noise tests. The noise samples displayed in the list can be found in the "fx-preset" folder.

Wizard: Here you can open the **"Noise print assistant"** dialog to help you extract a noise sample. You can find out more about this assistant below. (view page 833)

DeNoiser SE – Get noise samples from the „Effects“ menu tab.

1. Select an area in the selected Wave project or an object in a virtual project, where only the noise is audible. As a rule, you will usually get better effects with longer noise samples. If the length of the noise samples exceeds a minute, only marginal improvements can be obtained.
2. Select "Effects -> Restoration -> Get noise sample".
3. Select the range of the original sample in the wave project to which de-hissing should be applied or select the corresponding object in the VIP.
4. Open the DeNoiser SE via the "Effects menu -> Restoration -> DeNoiser SE".
5. If the sample selection list for the noise sample (under "Noise Sample -> File") is not already set to "Noise Print", rearrange the list entry accordingly.
6. Press the "Play/Stop" button for the real-time preview function. Once you are satisfied with the result, press "OK". Otherwise, change the parameters.

DeNoiser SE – Noise Print Assistant



Proceed as follows to generate a noise sample with the Noise Print Assistant.

1. Determine noise print length

Range length or internal standard value: If a range has been stretched, a range length will be used. You should be careful not to select an area longer than necessary, because each Noise Sample used in a real-time DeNoiser instance is added to the project. If however no range is selected an internal length will be used, which normally produces a useful result.

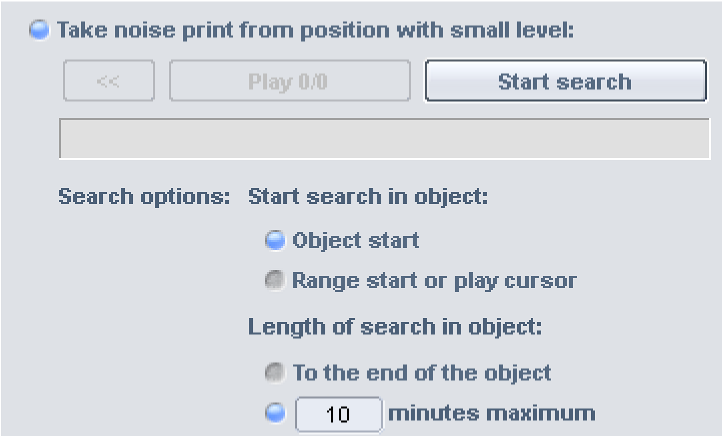
Target length: Set an explicit length for the noise sample.

2. Extract noise print from the audio material

Extraction at beginning of range or play cursor: The noise sample will be extracted from the range available, or otherwise the start position will be determined by the play cursor.

Extraction at the position of the lowest level: This search function analyzes the audio object for soft areas. Set the search area above the search options and click the "**Start search**" button. You can cancel the search by pressing "Esc".

After at least one quiet range has been found, you can select it using **<<** and **>>** and listen to it using "**Play...**"



Search options

Start the search in object – object type: The search area starts at object type

Start the search in object – area start or the play cursor: Search area starts at the area start, if you have defined an area previously. Otherwise the search starts at the play cursor.

Length of search in the object – until object end: The search for soft passages stretches until object end.

Search length in object – x minutes maximum: Enter the search length in minutes here.

3. Exit the assistant by clicking "OK".

Get Noise Sample

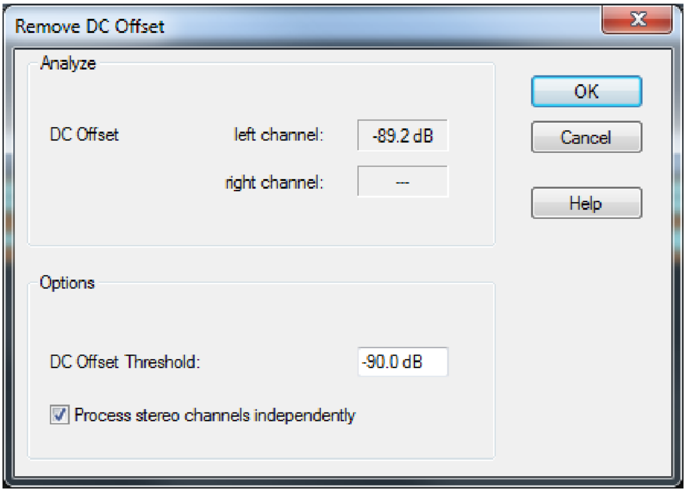
With this command you can create a noise sample. To do so, you must select an area in a Wave project or an object in a virtual project, which contains only the noise to be removed. The noise sample offers noise reduction a preset to aid in noise removal.

eFX_DeEsser

Detailed information on eFX_DeEsser can be found at "essential FX > DeEsser (view page 864)".

Remove DC offset (offline)

This function removes the DC offset of an audio file/object. This is useful if the sound card overlays a recording of your sample with a constant DC offset, which may cause clicks during cutting or editing.



Options: Here you can enter a minimum DC offset threshold, which would indicate where DC offset removal will kick in. If you unclick the corresponding check, stereo channels will be edited together. This serves to save computational time, especially when working with long files.

Stereo/Phase

Switch Channels

Use this function to swap the right and left channel of stereo samples in order to correct accidentally swapped channels when recording.

This function may be reversed, e.g. if you don't re-select the range, opening it again will restore the original material.

Multiband stereo enhancer

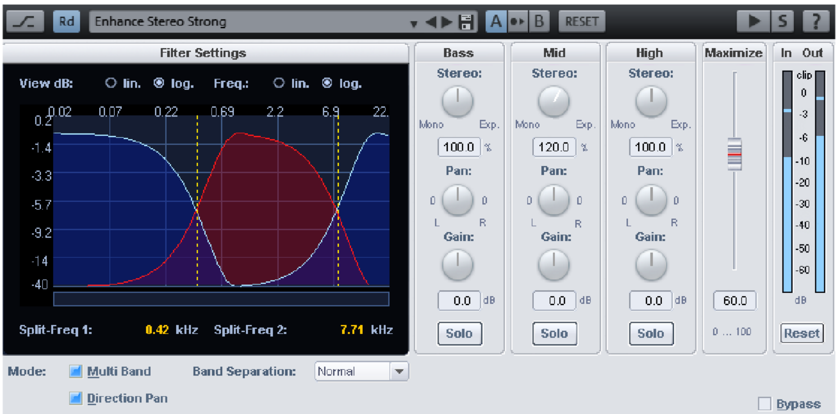
The multiband stereo enhancer performs detailed modifications and corrections to the stereo image in three independent frequency bands.

Thanks to the FIR complement filter technique, no discoloration of the frequency response will be introduced.

Using multiple bands to affect the stereo image has many important advantages over standard solutions without frequency separation. Increasing the width of the mid-range, for example, prevents "typical" problems like muddy highs and lows that result from removal. The reduction of mono compatibility caused by an increase of the base width is also limited to treatment of a specific band.

Important Multiband Stereo Enhancer applications include:

- Reduction or expansion of the stereo base width.
- More powerful bass by reducing the base width in the bass range.
- Control and correction of problems in the stereo image of a completed mix.
- Moving the mono section of a stereo recording in panorama (direction mix). This way, centered vocal recordings may be moved to the left or right of the stereo balance later.
- Damping or removing mono signals in the mid-frequency range in order to create room in tracks or drum loops. This allows the addition of further instruments or vocals.

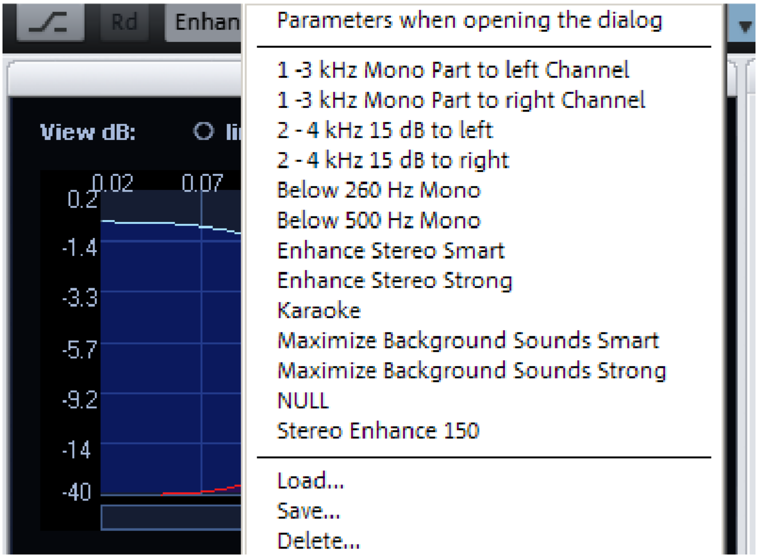


Overview

Filter presets are found at the top left dialog window.

The graphic displays approximately the frequency response of individual bands. The left axis label displays damping in -dB, the top axis label - frequencies in kHz.

General controls



By selecting"**Parameters when opening the dialog**", you will undo all changes made in the dialog since it was opened. By closing the dialog, you will apply all current settings.

Save, **load** and **delete** functions are integrated into the presets list, where they are available in the lower section. The default file extension is: *.ste

Bass, mids, and highs: You can set three frequency ranges for stereo, panorama, and gain (see below).

Maximize: Use this controller to compress the side-chain signal, which also increases the stereo transparency without influencing the mono compatibility. In "Multiband" mode, this setting affects the middle bands.

Solo mode: In solo mode, you can listed to frequency bands individually. This greatly simplifies the process of changing filter parameter settings. As an example, you can isolate a specific frequency range of a mix, and change the aspect of the stereo image for the range.

Multiband mode: The algorithm works with three frequency bands if this option is checked. The "Multiband" mode is not automatically active when using the enhancer in the mixer master section. This reduces the initial load on the CPU when selected in the master section.

Direction pan mode: This switches between two panorama control modes:

If the option is checked, only the mono portion (the center signal) is considered when changing the panorama setting. The panorama control functions as a directional mix controller. A centered vocals recording can thereby later be moved to the left or right of the stereo balance. The difference containing the sound sources allocated outside of the centre, remain unchanged.

In the inactive state, the panorama controls work as usual. The complete stereo signal (mono portion and the difference) is altered.

Phasecorrel. (Phase correlator): This open the phase correlator. This option is very useful when Solo mode is switched on: The base width and panorama setting for each frequency band can be checked visually.

Parameters for stereo manipulation

The following is available for each frequency band (bass, mids, highs):

Base width controller: Set the base width between 0 and 200. 0 indicates "Mono", 100 corresponds with unchanged base width (Stereo), and a value of 200 produces the maximum base width (difference signal).

Depending on the correlation between left and right, the level may increase when reducing the base width. In more extreme cases, this may cause maximum correlation, which occurs when the left and right channels are identical. When the base width is set to zero (mono), a level increase of 3 dB will occur.

Raising the bandwidth (values over 100) diminishes the mono compatibility. If the base width is reduced, mono compatibility will be maintained.

Panorama controller: Adjust the individual channels of the panorama here. Dampening for left and right is displayed above the controller in dB.

If the direction pan mode is active, the controllers operate as direction mixers. In this case, only the mono component (mid signal) will be included.

Filter setting parameters

Separation frequencies: Use these two sliders ("1" and "2") to adjust the separation frequencies of the three filter bands. The values are displayed in kHz (these correspond with the intersection of the neighboring frequency curves).

Band separation: This parameter influences different filter properties to increase precision. Select between "Low", "Normal", and "High".

Increasing the band separation setting has the following effect:

- The edge steepness of the filter curves increases and the transition range between two bands decreases.
- Dampening in the cut-off range increases (low setting: approx. 25-35 dB, normal: approx. 35-45 dB, high approx. 55-75 dB).
- The ripple of the frequency range of the bands decreases. However, this is not a problem, since the ripples of the individual filter bands compensate each other after combination, thanks to the complementary filter technology. In any case, the output signal will not contain ripples.

Multiband Stereo Enhancer in the master area of the mixer:

Access the stereo enhancer in the master area of the mixer by pressing the "**StEn**" button, which is located above the master faders. Right-click this button to open the stereo enhancer dialog window. The knob in the mixer is connected to the base width controller of the mid band.

If Multiband Mode is not active in the dialog, the changes affect the entire signal. In this case the knob functions as a standard base width control.

Stereo Enhancer - Tips and Tricks

General info concerning base width settings and standard use of the multiband stereo enhancer

- The human ear is generally incapable of localizing frequencies below 300 Hz. Stereo effects in the bass range can be disruptive as different delay levels often result in cancellation, which becomes audible in spongy and squishy playback of the basses.
- The treble range may be important for directional ; however, spreading the base width often causes annoying side effects in this case.
- For these reasons, the base width is usually enlarged or the direction of a mono source is changed in a middle band.
- In the bass area, the base width is set to mono by default. Enlarging the base width is only useful for special applications.
- The treble band is maintained by the following standard applications and is not changed.

Increasing the stereo base width

For this task, set the basic width controller of the middle band to a higher value (between 101 and 200).

More powerful bass by reducing the base width in the bass range

Set the base width controller of the bass band to mono. The setting of the lower separation frequency is also important for this task. Typical settings for this application are between 300 and 600 Hz. If the separation frequencies are higher, narrowing in the stereograph will already be audible, depending on the properties of the audio material.

Checking and correcting a completed mix in the stereograph

To check and possibly correct the base width and the panorama setting in various frequency ranges, proceed for all three bands as follows:

1. Activate "Solo" mode.
2. Open the phase correlator during real-time preview.
3. The base width may be estimated according to the amplitude of the display. The horizontal amplitude increases in relation to the vertical amplitude as the base width increases. The panorama setting may be verified by the gradient of the display. The display tilts to the side of the stereo channel with the higher level.
4. If necessary, correct each channel with the panorama and base width controllers.

Positioning a mono source in the panorama (mix direction)

The first step is to "filter out" the mono source (for example, vocals) so that the remaining part of the mix remains unaffected.

1. Activate "Solo" mode in this case as well and select the solo switch for the mid band.
2. Change both separation frequencies so that they just about pick up the source.
3. Now activate "Direction pan" mode so that the panorama controller acts as a direction mixer.
4. With the pan controller of the middle band, you may now position the mono source. The differential signal is retained.
5. Deactivate "Solo" mode to hear the effect of this frequency-selective directional mix on the overall signal.

Dampening or removing the mono signal from the middle frequency range

For instance, if you want to create space for adding vocals or solo instruments in the middle range of playbacks and drum loops, cancel out or dampen the original mid signal in the mid range by enlarging the base width.

As described above, attempt to "filter" out the mid signal well in "Solo" mode. The base width should then be set to the highest value (200) in the mid band.

Invert Phase

This function inverts the sample data within the selected range along the amplitude axis, i.e. the phase is inverted. Negative values are reversed to become positive values.

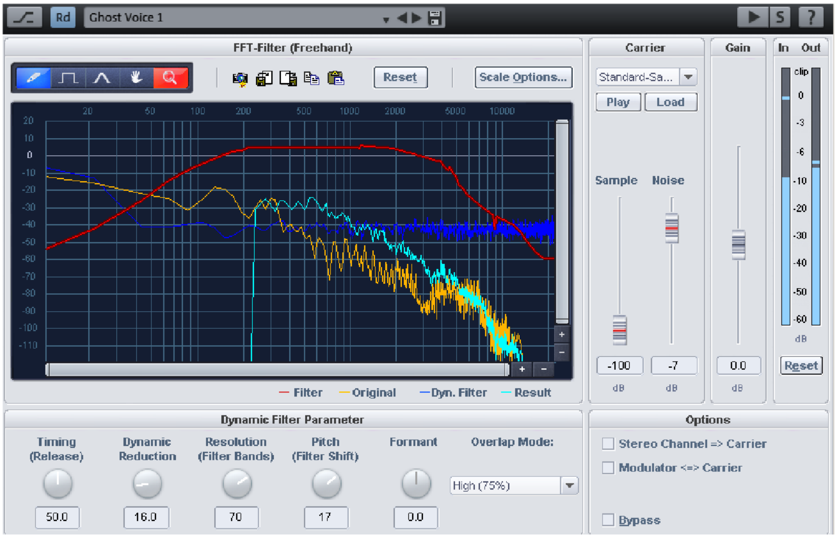
This way enables customization of samples with differing phase lengths. The function may be reversed and applied using both channels, just the left, or just the right in wave projects.

eFX_Tremolo Pan

Detailed information on eFX_Tube Stage can be found at "essential FX > Tremolo Pan (view page 865)".

Modulation/Special

Vocoder



The vocoder sends a carrier signal (e.g. a string instrument or a synth chord) through a modulator (e.g. speech or a song, or even drum loops) to make it seem as if the string instrument is speaking or singing.

This is achieved by transmitting the modulator's frequency characteristics to the carrier. The modulator signal is divided into a number of frequency bands and apportioned positions at regular intervals in the respective frequency bands. These measurements control a filter for the carrier that correspond to the same frequency bands.

Strictly speaking, a vocoder has two inputs and an output. Since Samplitude effects normally only feature one input, the carrier signal will be extracted within the effect to make it mixable with the white noise and any wave file. The carrier and modulator may also be swapped in order to use the vocoder input as the carrier.

The vocoder also features a real-time FFT filter for editing the vocoder signal.

Filter curves

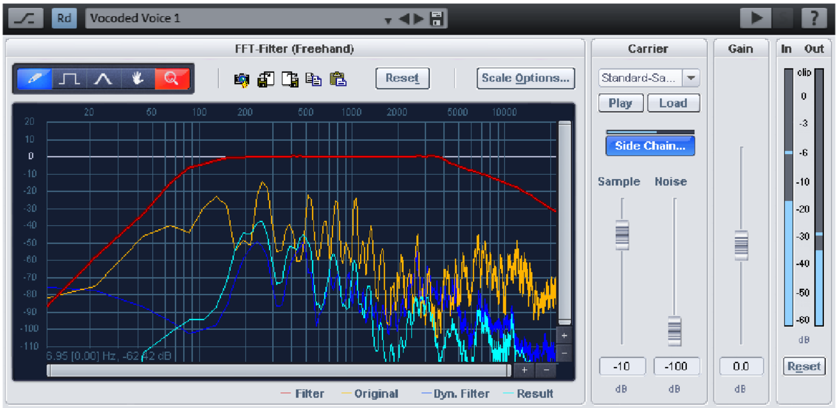
The yellow line corresponds to the frequency progression of the modulator, and the light blue line represents the carrier signal. The red line is a freely-adjustable FFT filter. The frequency progression may be drawn to optimize the vocoder results. The dark blue line is the resulting filter curve of the vocoder.

Reset: This resets the red filter curve to its initial state.

Carrier signal

Sample selection list: The desired carrier samples may be selected here. All open wave projects appear in the selection list as well as some special carrier samples that are loaded from the "Vocoder" directory. Preferred carrier samples consist of material with even frequencies e.g. orchestral chords, broad synthesizer sweeps, sound of the wind, etc.

Sidechain: The sidechaining option is now available for the vocoder if the effect is used **as a track or master effect** and the track isn't the Surround master.



Sample (dB): Use this fader to adjust the proportion of the carrier sample.

Noise (dB): This controller mixes white noise into the carrier. This is especially useful if the carrier material can't be modulated well or sounds too uneven. Whispering voices may also be produced in this fashion.

Volume (dB): Adjusts the vocoder output level.

Vocoder - options

Modulator <=> carrier: Swaps modulator and carrier signals. This is particularly useful if the "stereo channel as carrier" option is used.

Stereo channel as carrier: If this option is active, the sample from the selection list will no longer be used as a carrier signal, but an input signal channel will be used instead. The other channel will continue as the modulator.

This results in a more accurate synchronization of the carrier and the modulation signal, which is independent from play start.

Bypass: The "Bypass" button plays the original signal.

Dynamic filter

Response time (release): Affects the speed of the dynamic filter adjustment to the modulator spectrum. As the release value increases, the Vocoder follows the modulator slower and slower, and the sound changes sound softer and feature more reverb in the carrier. To improve the clarity of spoken words, this parameter should be set to a low value.

Dynamic (reduction): This parameter affects the dynamics of the modulator signal for reducing the modulation depth of the dynamic filter.

This prevents two often undesired side effects of modulation: on the one hand, the volume change of the modulator signal is added to the output signal in a slightly more moderate form, which may improve the power of the Vocoder voice. On the other hand, the low-level portions of the modulator signal are ignored in order to prevent modulation of the carrier by breathing or noise.

Instead of using dynamic reduction within the Vocoder, you can adjust the modulator signal dynamics (or the resulting signals) with the dynamic tools integrated in Samplitude.

Resolution (filter bands): Dynamic filter resolution (approximately dependant upon the number of filter bands). The best results are achieved with medium to high rates.

Pitch (filter shift): The dynamic filter of the Vocoder shifts the frequency up or down to create certain pitch effects. For the clearest speech results, it is recommended not to change this parameter.

Formant: Stretches the dynamic filter curves to manipulate the formants. This changes the characteristics of the Vocoder voice.

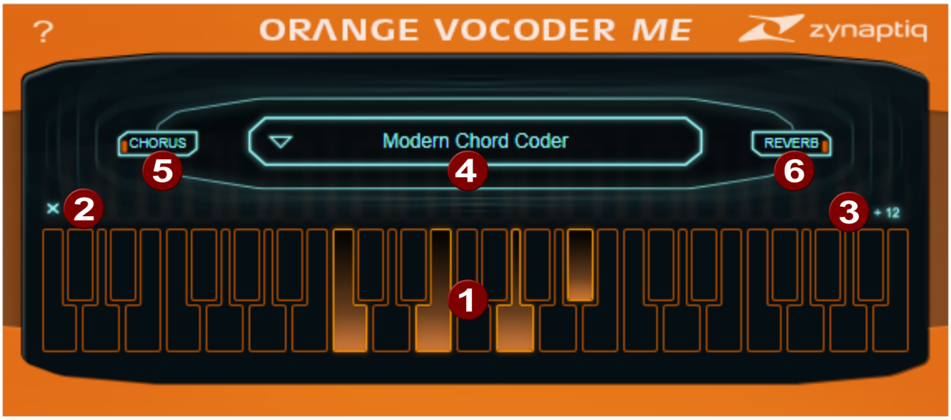
Overlapping: This internal parameter changes the overlapping of the time window for the modulator signal spectrum calculation.

On the "low" setting, the Vocoder sounds softer and more melodic, but some of the clarity of speech may be lost.

Orange Vocoder ME

The Orange Vocoder MAGIX Edition from zynaptiq is a resurrected form of the legendary Prosoniq Orange Vocoder and is available exclusively for MAGIX as a limited ME version for PC. What you get is the best-sounding software emulation of an analog vocoder available on the market.

A vocoder analyzes the frequency components of an input signal (modulator) and uses this to modulate a carrier signal. What this means is that the sound of the carrier signal is filtered using a number of bandpass filters so that the frequencies are either strengthened or weakened according to the frequency distribution in the modulator. For example, you could use a human voice as a modulator signal in order to make a synthesizer (the carrier signal) talk. Additionally, rhythmic pads can be created by modulating an area with a drum loop.



In the Orange Vocoder, the carrier signal is generated with a synthesizer. You can set the notes for the carrier signal via the keyboard (**1**). The x to the left (**2**) deactivates all the keys. +12 (**3**) makes the synthesizer sound an octave higher.

The drop-down menu in the center (**4**) can be used to select a preset for the sound of the vocoder. Eight presets come included. You can purchase an additional two sets containing 25 presets each in the MAGIX Shop.

The sound of the vocoder can be further refined using chorus (**5**) and reverb (**6**).

> Tip: You can play different chords with the Orange Vocoder ME by loading it as an object effect on your vocal sample. Then, split the object at the chord changes and use the vocoder keyboard to set the chords you want in each object.

eFX_Chorus/Flanger

Detailed information on eFX_Chorus/Flanger can be found at "essential FX > Chorus/Flanger (view page 851)".

eFX_Phaser

Detailed information on eFX_Phaser can be found at "essential FX > Phaser (view page 852)".

eFX_Vocal Strip

Detailed information on eFX_Vocal Strip can be found at "essential FX > Vocal Strip (view page 866)".

Corvex

More information is available in "Effects -> MAGIX plug-ins (view page 849)".

Backward

With this function, the data of the sample in the wave project within the selected area/the selected object in the virtual project along the time axis is reversed so that it can be played from back to front.

Encoder Preview

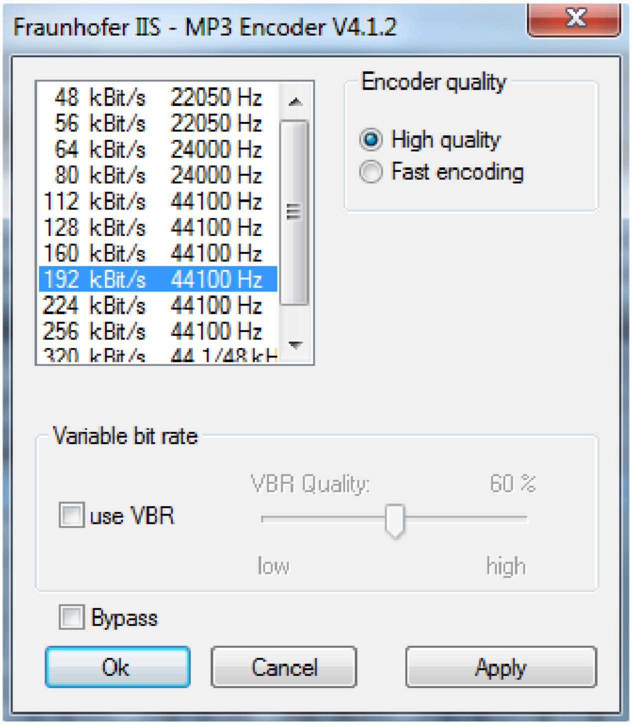
MP3 Preview in the Mixer Section > Master Plug-ins

Only in Samplitude Pro X4 Suite

MP3 Preview Plug-in: In the master plug-ins section of the mixer there is a preview function available which can be used to prepare MP3s or AAC files for export. This makes it possible to hear in realtime what the generated MP3 file will sound like after encoding.

During the mastering process you have the option of taking the idiosyncrasies of the encoder into account and adjusting the quality relative to the requirements.

You can change the export settings "Bit Rate" and "Encoder Quality" directly in the dialog window.



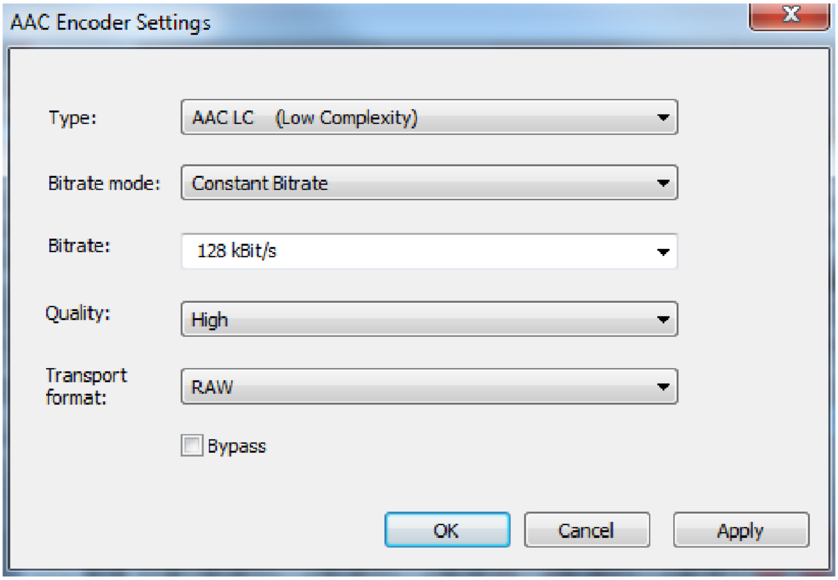
AAC Preview in the Mixer Section > Master Plug-ins

Nur in Samplitude Pro X4 Suite

AAC Preview Plug-in: In the master plug-ins section of the mixer there is a preview function available which can be used to prepare AAC files for export. This makes it possible to hear in realtime what the generated AAC file will sound like after encoding.

During the mastering process you have the option of taking the idiosyncrasies of the encoder into account and adjusting the quality relative to the requirements (e. g. „Mastered for iTunes").

You can change the export settings "Type", "Bit Rate Mode", "Bit Rate", "Quality" and "Transport Format" directly in the dialog window.



MIDI Velocity Dynamics

MIDI Velocity Dynamics Settings...

Detailed information about MIDI velocity dynamics settings is available in "MIDI editors -> Velocity dynamics MIDI functions (view page 338)".

Apply MIDI Velocity Dynamics

This command applies the current MIDI velocity dynamics settings to MIDI objects.

Set MIDI Velocity to Fixed Value

This command applies the current MIDI velocity dynamics settings next to the "Set value" button to selected MIDI objects.

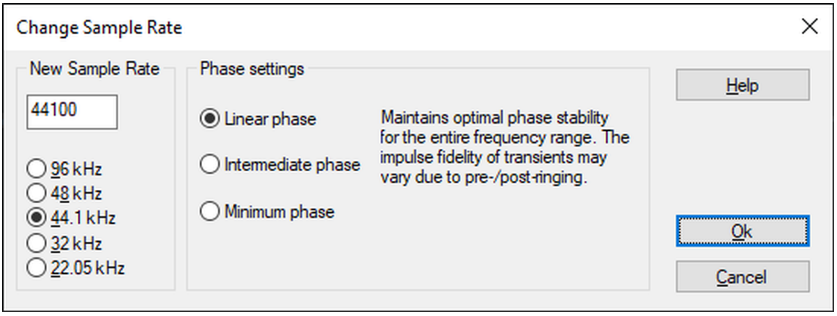
Randomize MIDI Velocity

This command applies randomly varied MIDI velocity dynamics to selected MIDI objects.

Sample manipulation

Adjust Sample Rate (offline)

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



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

If you reduce the sample rate, on the other hand, there will be a loss of overtones. When you halve the sample rate of a 44.1 kHz sample to 22.05 kHz, the frequency response of the resulting signal is limited to 11.025 kHz. The frequency response always corresponds to half of the sample rate.

By clicking on "OK", you may enter the file name for the new project.

Phase settings

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.

Note: You can carry out the sample rate adjustment while recording and playing. The quality of the resampling used in this case can be set in "File > Program Preferences > Resampling Quality Options... (view page 640)".

Backward

With this function, the data of the sample in the wave project within the selected area/the selected object in the virtual project along the time axis is reversed so that it can be played from back to front.

Makes Loops (offline)

With this function you can call up a complex algorithm for loop editing in wave projects.

First, select a range that roughly shows the loop positions. If you have selected "Loop" in the transport window and move or edit the range on playback, you'll be able to find some good loop points. Select the "Range menu > Split range" view. Here, the sample is displayed in three sections.

"Build Physical Loop" allows the borders of the range top be set to the precise crossover point of the sample in order to get loops without any crackling. Moreover a crossfade of the audio material at the loop end is performed with the audio material from the loop beginning, to ensure a smooth transition.

If a marker is placed before the highlighted range, the range between the marker and range start will be used for the crossfade. Therefore you can also manipulate the length of the crossfade, like when sampling. If you move the marker close to the start of the range, you will get short crossfades, if the marker is placed far away from the start of the range, long crossfades will be the result.

For a crossfade you should bear in mind that the distance between the marker and the range start cannot exceed the length of the range itself.

Plug-ins...

> You can find more information about VST plug-ins in "Mixer > Effect Routing / Plug-ins Dialog -> Effect Sequence / Plug-ins".

Common controls

Parameter smoothing/ Controller knobs

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

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

Essential FX Suite

The essentialFX Plug-in Suite offers high-quality audio tools based on state-of-the-art algorithms. They have been designed with special emphasis on a clean and simple display of the individual parameters.

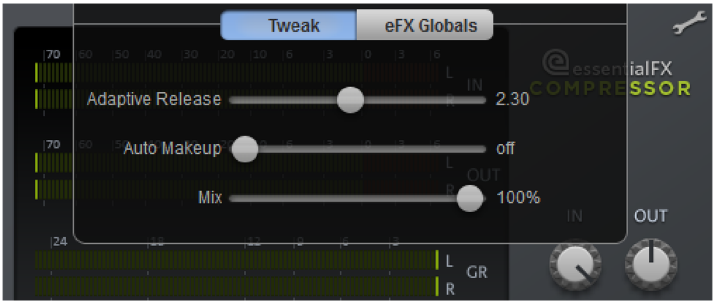
The uniform control concept combined with low CPU resource demands makes this effects suite the perfect tool for a wide range of studio tasks.

essentialFX Settings

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

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

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



eFX Globals

These settings apply to all Essential FX.

Metering: Here you can set the meter display.

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

Mouse: Here you can set the display reaction to mouse movements in the essentialFX graphic display.

- **Linear mode** lets the faders move in response to vertical and horizontal mouse movements.
- **Circular mode** lets the faders be controlled by circular mouse movements.
- **Mode As Host** lets the faders move in response to the mouse exactly how mouse settings in the host program specify it.

- **[Shift] Factor** specifies the factor for fine adjustment of individual faders with the mouse with simultaneously held Shift key.

Chorus/Flanger

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

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

Chorus flanger parameters

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

mode:

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

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

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

- **Rate:** This specifies the speed of the modulation. Lower rates provide slight hovering effects, and high speeds produce a wobbling, typically distorted “underwater” sound.
- **Depth:** This parameter specifies the depth of the modulation, i. e. the maximum deviation of the modulation and the resulting pitch bending.
- **phase:** This fader moves the right channel's oscillator phasing relative to the left, wave is put back to the right. So that the Tremolo effect drifts apart in the stereo field with ever increasing values. At 180° both oscillators work in reverse, therefore the stereo effect is at its strongest.
- **Feedback:** This parameter defines the portion of the delay that is sent back to the input. Feedback causes the effects of modulation to be more drastic and cutting. Nullification of the feedback is set at the middle of the fader. Set to the right, the feedback is fed to the input equi-phasal; to the left, the feedback occurs. Both variants may sound very different depending on the signal, since they prefer different frequency ranges for dissonance.
- **Mix:** Regulates the mix ratio of the original signal and the delayed portion.

Tweak

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

Phaser

The phaser is often mistaken for a flanger due to its typically sharper and cutting effect. In any case, the pitch is not modulated. Instead, the modulation process burrows multiple notches into the frequency response, somewhat like a comb filter. Just like an airplane taking off, the phaser functions with a similar jet effect. It is suitable for enduring signals like synth surfaces or for producing sound designs to create atmosphere or distortion effects.

Phaser parameters

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

Mode: The selection includes a number of filter stages. At **4 stages/8 stages**, a more plastic effect is achieved, and more complex patterns are reached at **16 stages**. Please note that the more stages are involved, the more computing time will be needed.

- **Rate:** Speed of filter modulation. The essential effect is the same for both chorus and flanger.
- **Depth:** Similar to chorus/flanger, whereby it's the filter notches that are addressed, and not the pitch modulation.
- **phase:** This fader moves the right channel's oscillator phasing relative to the left, wave is put back to the right. So that the Tremolo effect drifts apart in the stereo field with ever increasing values. At 180° both oscillators work in reverse, therefore the stereo effect is at its strongest.
- **Feedback:** The feedback portion produces a more drastic effect in this case. Similarly to the chorus/flanger, co-phasal or opposite-phase feedback is possible.
- **Mix:** Regulates the mix ratio of the original signal and the delayed portion.

Tweak

- **Center Freq:** Sets the mid-frequencies of the phaser. The filter modulation acts on these frequencies.

Reverb

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

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

Reverb parameters

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

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

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

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

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

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

- **Size:** The size of the room being simulated or of the reverb plate. This setting has a direct effect on the distance of the reflection and indirectly on the resonance behavior: smaller rooms/reverb plates possess a larger inclination for natural resonance than larger ones.
- **Time:** Length of reverberation time (in seconds). This value refers to the RT60 time, i. e. how long the reverberation requires for the level to fall 60 dB.
- **Damping:** Real rooms and reverb plates dampen certain frequencies according to their construction. This is most often audible in upper frequencies. Using this

parameter you can set the cut-off frequency, above which the reverb will be particularly strong. In some cases, the "Time" parameter should be adjusted, since sometimes the subjectively perceived reverberation time will also be affected.

- **Modulation:** For several critical signals, the reverberation may produce disturbing resonance. This can be scattered by temporal modulation of the delay signal used in an algorithm. Excessive application may create an effect similar to chorus in tonal signals. The liveliness of the reverberations will be emphasized with lower values. In this case, the modulation applies a random pattern, the intensity of which may be visualized via the "MOD" display.
- **Mix:** This regulates the ratio of the unprocessed signal to the portion including reverberation.

Tweak

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

Stereo Delay

The stereo delay is a simple too for typical bread and butter delay effects. The stereo delay offers the "analog algorithm," which features the sound of echo devices of yesteryear.

Stereo delay parameters

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

Mode: This selects between the essential algorithms.

Digital: Normal, transparent delay

Analog tape: Analog tape delay simulation. In this mode, a band echo is simulated with a typical compression and saturation behavior, including phase shifts with high feedback settings.

Analog BBD: Simulation of a bucket brigade delay (BBD, bucket brigade delay). These devices, which originate from the pre-digital era, used analog building blocks for storage. The signal was held for a short time in a relatively simple circuit and then moved on to the next. This "bucket brigade" principle created a longer signal delay. BBDs create different delay times by varying the system beat (clock), meaning that for short delays the beat is faster, and for longer it is slower.

- **Tempo sync:** If this button is active, the plug-in is directed at the host/sequencer tempo. In this mode, changes are made to the delay period via the L/R delay using the musical snap grid (e. g. 1/4 note).
- **Delay L/Delay R:** Specify the delay period for the left and right channels here.
- **Damping:** This specifies the cut-off frequency at which the highs are dampened during the delay. This is useful for making the delays reverberate more naturally or for creating special effects (reggae/dub-style effects).
- **Feedback:** This parameter regulates the internal amplitude of the delayed signal that is fed back to the input. In "Digital" mode, this process is completely transparent; in "Analog," on the other hand, higher values, a very loud input signal, or the sum of these will make the use of dynamics compression audible. In both modes, the nullification of the feedback parameter is in the center of the fader. To the right, the plug-in works in "Dual delay" mode (both sides work independently), and to the left, "Ping pong" mode will be activated (the delayed signal alternates between the left and right sides).
- **Mix:** Regulates the mix ratio of the original signal and the delayed portion.

Tweak

- **Feedback Low Cut:** This control sets the filter frequency of a high-pass filter for the "Feedback." Signal components below this frequency will be filtered out.
- **Tape / BBD Noise:** Here you can control system noise for both analog modes. Especially if you work with high feedback rates, noise can create quick or stable oscillation, making the effect even more authentic.
- **BBD Stages:** Here you can determine the count of memory cells. For very long delays, chips with many memory cells are used. This explains why bucket brigade echoes with long delays sound so muffled and dirty. Please keep in mind that the shorter the delay, the faster is the "Virtual" system beat. For performance reasons, the system beat is limited. The limit is displayed below the delay control (with deactivated "tempo sync") as soon as it is reached, e. g. 46 ms **(min)**.

- **BBD Compander:** Here you can simulate Compander settings. Due to per-cell loss, BBDs have low system dynamics. For this reason, some have an integrated compander (compression at the input, counter-expansion at the output). Strong compander settings interact noticeably with the input signal, especially with high feedback, because the input effects the expansion ration at the output, even when there is no signal at the input.
- **BBD Clock Drift:** This parameter can add drift (during audible jitter effects) by slightly varying the system beat of BBD cells. It works similarly to an LFO, but is randomized.

Compressor

This plug-in is a simple but effective tool for reducing the dynamics of a signal. Percussion tracks may be modeled more compactly and with more pumping, vocals may be integrated into the mix better, or the entire sum or group or signals may be made denser.

In contrast to some other plug-ins, this type works the compressor with a comparatively soft curve and an adaptable regulation process, it therefore compresses completely musically. Additionally a separate input (external sidechain signal) can be defined as the regulation source.

Compressor Parameter

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

Internal sidechain corresponds with conventional methods; this selects the actual input signal to control the compression process.

If another track in the project triggers the compression, select **External sidechain** here. This will use a second stereo track routed to the plug-in. Make the regular adjustments in the arrangement to route the sidechain signals.

Note: Sidechaining is not available in some host programs.

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

- **SC filter:** This fader determines the application/middle frequencies, that are filtered with the control signal for level detection. The filter works selectively as a high-pass, band pass or low-pass. In many cases evaluating the detection is a good for optimizing the leveling process. So, for example "high-pass" filtering enables complex sources (drums or sum signals) to achieve consistent regulation without typical pumping artifacts, since the process is mainly activated via mids and highs.
- Using the filter type switch the properties of the detector circle can be drastically changed. The speaker symbol is there to preview changes.
- **Threshold:** This sets the response threshold for regulation. For example, -20 dB indicates that the input/sidechain signal is only compressed once it reaches -20 dB; below this level, no change or hardly any change will take place. Note that this plug-in deals with the threshold window independent of the program. It can be the case that a small regularization has already taken place at -25dB. This so-called soft-knee characteristic ensures a soft, musical compression process.
- **Ratio:** This parameter regulates the compression ratio. For example, a value of 10:1 means that when the threshold level is reached, the input level is increased by 10dB, while the output is only 1dB louder. A lower compression, e. g. 2:1, is recommended for subtle compression of group or sum signals, while 50:1 produces a very hard limiting that cannot be made inaudible by even the most transparent compressor.
- **Attack & release:** This parameter specifies how quickly the compressor responds when the threshold is reached (attack) or how quickly the signal applies normal amplitude once the signal drops below this level (release). The transient time settings can be defined in another area. Note that due to the adaptable plug-in leveling actual times may differ. This semi-automatic process favors quick adjustment, without leading to the often dreaded artifacts (rough sound image with rushed regulation, overly low/inefficient compression with long time constants).

Tweak

- **Adaptive release:** With increasing values, the compressor works increasingly with "Adaptive release". This means that the longer and harder the compressor intercedes upstream in the signal path, the longer is the resulting release phase.
- **Auto makeup gain**: Normally, you have to continuously adjust level reduction to generate "compression" at the same maximum level. This is done by

activating auto makeup gain. The volume difference expected from the set working parameters is determined and applied as an output factor after master regulation. If you prefer to adjust the "classic" level reduction and amplification manually, you can deactivate this function.

- **Mix:** Addition of the original signals is done to get transients and spectral balance form the source. A "mixed" signal is particularly discreet, more transparent, and less "squishy" with vocals, whereby the compressed part usually has a higher level reduction than without adding the original.

Gate

The Gate is operated in much the same way as the eFX_Compressor. In particular, the side-chain function, and selecting various filter types can be also be applied here.

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

Gate parameter

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

sidechain int/ext: As with eFX_Compressor an external control input can be used for triggering.

Note: Sidechaining is not available in some host programs.

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

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

- **sc filter:** The control corresponds to the essentialFX Compressor.
- **threshold:** This fader defines the threshold beneath which the gate should be initiated.
- **range:** Here you can set the Gating strength. Right extreme means that the signal beneath the threshold is cut out completely. Subtly attenuating the signal can remove background noises or breathing from vocal tracks. Where necessary the process can be attenuated even further by switching on the soft knee mode.
- **attack:** Regulates the attack time from the closed gate to the point at which signal is let through again.
- **release:** Sets the time, that the gate requires to go from a normal state to gating.

Tweak

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

Limiter

This plug-in is a simple yet effective tool for increasing the loudness of your audio signal. This creates a compressed but loud signal without allowing the defined output volume to be exceeded.

The limiter operates with a soft curve and an adaptive regulator that ensures that the sound remains absolutely musical. You can also use a separate input (external sidechain signal) to define the source for the regulation.

Limiter Parameters

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

- **Threshold:** This sets the threshold; the effect will be applied above this level.
- **Release:** This sets the time frame between the point when the signal drops below the threshold and the complete reduction of the effect.
- **Clip Gain:** This sets the amplification factor.

Tweak

- **Attack:** This sets the time frame between the point when the signal exceeds the threshold and the maximum extent of the effect.
- **Adaptive Release:** With increasing values, the limiter works increasingly with "Adaptive Release". This means that the release phase gets longer according to how long and hard the limiter has been applied to the signal processing.
- **Peak vs. RMS:** Here you can choose whether to display the eFX_Limiter peak levels (slider completely to the left) or the RMS levels (slider completely to the right). You can also set the display between peak level and RMS values to any ratio you like.

- **Auto Makeup Gain:** Here you can set the automatic tracking of the output amplification during level reduction in order to maintain compression at the same maximum level. The volume difference expected from the set working parameters is determined and applied as an output factor after master regulation. If you prefer to adjust the level reduction and amplification manually, you can deactivate this function.
- **Input Low-Cut:** You can use the fader to set the filter frequency of a high-pass filter for the input signal. Signal components below this frequency will be filtered out.
- **SC Low-Cut:** This sets the filter frequency of a high-pass filter for the sidechain signal. Signal components below this frequency will be filtered out.
- **Clip Shape:** The left velocity of the fader (soft) corresponds to the oscillator sine wave. The oscillator signal is gradually morphed to the right in the direction of the rectangular wave. This makes the effect more intense, i.e. "harder".

Tube Stage

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

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

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

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

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

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

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

Tube Stage Parameter

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

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

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

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

- **pre-eq:** This knob (called "Tilt" or "Level" on some devices) controls the effect of a soft, passive 6 dB filter for prefiltering signals before the (first) tube stage. If turned to the left the bass is emphasized and the highs are dampened. To the right the inverted filtering happens. With this filter the signal can be correspondingly processed before "Warming up" by selecting the prominent of desired section. Compared to a standard EQ the effect of this filter is more subtle, but it does possess a high 'musicality', by the internal circuitry and also because of the slightly different phase response.

- **gain:** This set the entire amplification factor. If two two tube stages are selected using 'stages', the available Gain is divided by two and is distributed equally over both stages.
- **post-eq:** Operation and function is like pre-eq, however, this filter is placed behind the (second) stage tube.

Tweak

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

DeEsser

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

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

DeEsser Parameters

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

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

Tweak

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

Tremolo Pan

This plug-in is on one hand a Tremolo, that creates typical textures for synths or vintage guitars. On the other hand you can create interesting, rhythmic stereo effects with auto panning.

Tremolo Pan Parameter

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

rate: Sets the speed of the internal oscillator in Hz or note value (see "tempo sync").

shape: The left velocity of the fader corresponds to an oscillator's sine wave. The oscillator signal is gradually morphed to the right in the direction of the rectangular wave. This makes the effect sound intense, "harder".

phase: This fader moves the right channel's oscillator phasing relative to the left, wave is put back to the right. So that the Tremolo effect drifts apart in the stereo field with ever increasing values. At 180° both oscillators work in reverse, therefore the stereo effect is at its strongest.

depth: This determines the strength that the oscillator signal has on the Tremolo/Panning effects.

tempo sync: when the switch is activated the "rate" fader snaps to values on the musical grid.

Vocal Strip

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

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

Vocal Strip Parameter

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

- **highpass:** This fader determines the cut-off frequency for a steeply sloped highpass filter (24dB/octave) to remove part with low frequencies, such as rumbling or noises created when touching the microphone.

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

Tweak

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

> **Note:** Detailed information about compressor-specific settings can be found under “EssentialFX > Compressor > Compressor Parameter (view page 857).”

Vintage FX: CORVEX – Chorus/Flanger

When referring to the chorus and the flanger effects in CORVEX, we are not dealing with just one of the two effects, but also the variations between them that go beyond the standard concept.

A chorus generates the typical "floating" sound that is commonly used for guitar sounds or synthesizer pads. You can add acoustic "depth" to an instrument and make it sound "fuller" or even create the illusion that several of these instruments are playing simultaneously.

The chorus sound is created with the "Doppler effect". You've probably noticed this phenomenon in everyday life when the siren of an approaching ambulance sounds more high-pitched the closer it gets and then sounds more low-pitched as it drives further away. This effect is a result of the speed of the sound which first increases and then decreases, thus also changing the sound pitch. The wave length also changes relative to the frequency. If there were a second siren at your location, an oscillation would develop between both sounds (just like when two instruments are out of tune).

Chorus also splits the signal into at least two components: a direct sound and an effects portion and there are multiple effects components in CORVEX.

The Doppler effect is created by a short signal delay from the effect. For most devices, this delay is within the range of 10-30 ms (as in this one) This means that it is short enough that it won't be perceived as an echo. The times would also be similarly short if you duplicated a guitar track, for instance. A short delay in the mix with the direct signal already sounds "doubled", but this is not authentic. This is where the above-mentioned "out-of-tune" effect comes in: The pitch of the effect signal is slightly modulated by gently "drifting" forward and backward in the delay curve. This creates a modulation with a liveliness that is influenced by the speed of this "drift".

The "Flange" effect is similar to that of the chorus, but it has a different technical and historical background. It resulted by chance: Someone (various sources say John Lennon) slowed down one of two running interconnected tape machines in a studio with his hand. The result was a rather brief delay of the second signal compared to the first, brought about cancellations within the frequency spectrum, leading to a comb filter effect (the sum of both signals creates "peaks" and "dips" in the spectrum that look familiar to the teeth of a comb).

Flanging is basically a chorus effect with a shorter delay time (less than 10 ms). Releasing or doubling the signals is not the main focus in this case. Here we are interested in a more creative distortion of the frequency response.

A complete flange effect will definitely require feedback: the flange portion is returned to the input to increase the effect. People often refer to this as the "jet effect" since it resembles a jet on take-off.

CORVEX – delay/modulation section

drift: This parameter introduces drift. It works similarly to an LFO, but is randomized.

time: Here you can specify the delay time and fundamentally set if you want to have a chorus or flanger sound. With the flanger, delay times are usually between 1 – 10 ms, as, with these times, the typical comb filter artifacts usually lie within the audible frequency area (Frequency = 1 / delay in ms). For a chorus, values of 20 - 40 ms are normal.

Wars of attrition are fought about what the actual correct delay times are. However, we don't advise you to strictly conform to such standards. Just let your ear decide.

voices: Use this to set how many internal voices the effect should contain. Two to max. eight delay units can be activated here. With more than two voices the sound becomes fuller and fatter. Uneven voices (1, 3, 5, 7) are assigned to the left channel, even voices (2, 4, 6, 8) to the right. The subsequent elements like the filter and diffusion unit also count as part of these voices/delay units. For this reason, you should note that an increase in the number of voices leads to increased CPU strain.

Span: The time of each delay unit activated by 'voices' can be shifted using this controller. Example: With the 'time' potentiometer you can set 10 ms and select four voices. "span" at 50% would mean that voice 2 would be delayed by 15ms, voice 3 by 20ms and voice 4 by 25ms. For example, you can break up the resonance by increasing the "span" value at high feedback rates or break apart the sound field at large stereo width.

The modulation depth: The depth or intensity can be adjusted here using the small fader below "time". Here, you'll see one of the most important properties of these three effects: the small fader on the lower row on the front plate is seen as having the same value. They are basically modulation targets. If, like in this case, the modulation target of the LFO is the "time" fader, you have set the depth of the pitch shift.

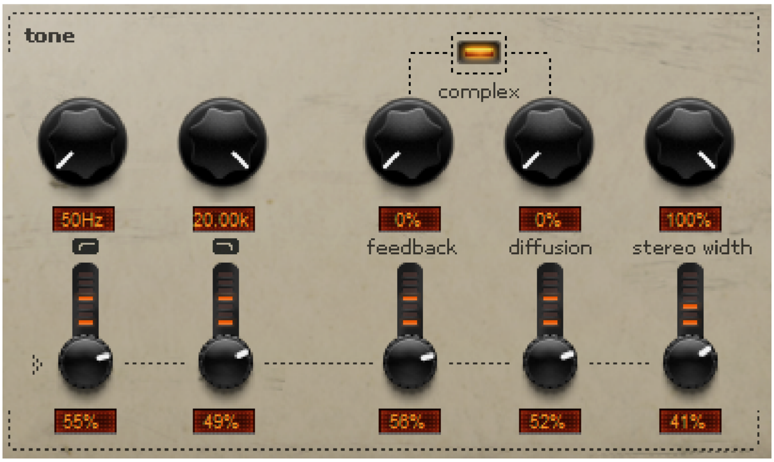
At max, the amplitude is at its largest, in the minimum position the effect remains static.

Modulation speed: Slow times create a quiet beat; high speeds sound like vibrato or, in extreme cases, like it's under water. The type of modulation can be selected using the buttons.

The **sync** button snaps the speed to the song tempo. Alternately you can set the modulation tempo in Hz manually.

Waveform selection: sine, square (sqre) and random (rand) are available. The sine wave is suitable for rhythmic, quiet sounds. On the other hand, a square wave sounds quite drastic and very rhythmic. Random mode is a good partner for ambient sounds or sound landscapes, the progressions are not foreseeable and can add an interesting accent to the sound.

CORVEX – Tone/Filter section



The **low and high cut filters** limit the signal within each voice/delay unit.

feedback: Here you can set how strongly the signal in each unit is fed back to itself. At high values you can get a typical, cutting flanger sound. What makes CORVEX stand out from many other digital devices is that high feedback rates do not result in overmodulation. In each delay unit, the signal is fed at a saturation level which brings about a soft, analog-sounding restriction.

diffusion: This is one of the most interesting parameters of CORVEX and rather unusual for this effect range. Normally, larger delays are audible as discreet echoes. Using "diffusion" softens the signal in each voice, scattering its shape. In extreme cases and with sufficiently high "time" values, even reverb-like sounds are produced. This enables a small room to be replicated using "time", "voices" & "span", and "diffusion" can be applied to simulate the natural properties of scattering signals over surfaces. With high feedback values, simply modulate the pitch a little (the small controller below "time") to break up the creation of comb artifacts resulting from static repetitions.

stereo width: Controls stereo width. You can widen the audio field using this parameter.

complex: Usually each voice pairing functions in "PingPong" mode, that is, the left channel is thrown back to the right and vice-versa. In "Complex" mode, the feedback & diffusion parameters are combined, which produces a quite chaotic sound. In "Complex" mode, each one of the 8 possible voices affects every other one, i. e. echo repetitions ensure that diffusion is even faster. Here, at the maximum number of voices, drastic spatial sounds are possible. For this reason, CORVEX manages to eclipse some specialized reverb effects...

CORVEX – modulation depth intensity setting

As mentioned above, this relates to an **intensity setting of the modulation depth**: Each of these faders specifies how much LFO affects the relevant parameter above it.

The following applies:

- The envelope of the LFO affects the "time" parameter directly, meaning that lighting of the round LED in the modulation section results in an increase to the delay times.
- The LFO also affects all other parameters to the right of the middle in a similarly direct fashion. To the left on the other hand, the inverse value of the modulation envelope is taken. If, for example, you have selected "sine" as the waveform, the inverse curve shape will correspond to a sine wave which is mirrored along the X-axis. In practice, this results in a temporal shift of modulation by exactly half of one period length.

Basically, the modulation values combine with the settings of the main pot. For example, to get a rhythmic stereo width introduction, set the stereo width control to 0%. Switch on "sync", adjust the "speed" to "1/4", and set the small fader for the modulation target (stereo width) to the right. This way the stereo picture plays at maximum every fourth note. Now turn the small pot below "stereo width" to the left: the stereo picture now plays on the off beats. This is where the inverse envelope is effective and the mentioned temporal shift is kept intact.

Use the exact same method to modify the other modulation targets and get sounds out of CORVEX that a normal chorus or flanger wouldn't know. We've created a few presets that make heavy use of modulation which can be used to quickly get things right...

Vintage FX: ECOX - Echo/Delay

This delay provides a creative playing style that is different from common delay effects.

The delay times can be changed during playback without creating scratchy digital artifacts. Instead, the times are faded out softly, similar to the old tape echo machines or bucket-brigade circuits that used tape speed to change the delay.

ECOX can reproduce this type of sound relatively easily including tracking fluctuations and loss of highs that always occur during each repetition (feedback).

Like the CORVEX, the internal feedback features a dual-band filter (low and high-cut) for creating dark, light, or mid repetitions (depending on the settings).

A special characteristic of ECOX is that the delay cannot be digitally distorted. Even in an endless "looped" repetition, the signal is not allowed to distort too much. Instead it is compressed by an increasingly slighter degree and distorted like a tape.

ECOX - delay time

drift: This parameter introduces drift. It works similarly to an LFO, but is randomized.

left + right delay time: The delay times can be adjusted separately for left and right. Use the "synced" settings to select the note value which the controller should snap to. The same note values as with CORVEX can be chosen. You can also leave out "sync" and specify free delay times in milliseconds.

The **sync** button beneath it affects the display of the delay time of the left and right channel in the bar grid.

Link button (lock symbol): Press this button to change both channels simultaneously with a delay knob. The link function also has an effect on the two modulation depth faders of this section.

Modulation depth left/right: These two knobs control the modulation depth and influence the pitch of the resulting delays.

ECOX - Modulation Section

The same features may be selected as with CORVEX. The same conditions apply as those mentioned above to change the pitch modulation.

ECOX - feedback path

Filter section, feedback, diffusion, stereo width: see CORVEX - tone/filter section (view page 870).

More information about about modulation depth can be found under CORVEX – intensity settings for modulation depth (view page 871).

Vintage FX: FILTOX - Multimode-Filter

Much like CORVEX and ECOX, FILTOX is also a "modulation effect". These settings have to do with frequency response bending. Modulation source controls **two filter units: filter left/filter right**.

Possible areas of application are synthesizer sounds (filter sweeps on pads) or creative distortions in drum loops (e. g. for variations, fills, etc). With guitars you can create typical 'wah' effects either by tempo modulation or in a special mode with modulation via the signal envelope curve.

The 'core' of FILTOX is a stereo multimode filter based on an analog model (Chamberlin 2-pole filter), which is known as an Oberheim filter. With FILTOX two of these modules are cascaded per channel to achieve a switchable 24dB roll-off.

Our digital model of the filter is designed so that it delivers the typical "analog" sound character, but it's also useful if you want to apply internal overmodulation. In this case, interaction occurs between the cut-off frequency and resonance, which makes the sound seem "undigital".

FILTOX – Parameters

The following filter types are available:

- Low-pass
- Band-pass
- Notch
- High-pass

Usually, these filter types are designed so that they are able to be toggled. However, a "state-variable" network is used with the Chamberlin filter, which means that all filter types can be taken from "taps" simultaneously. So why leave these taps static when they can be processed dynamically...?

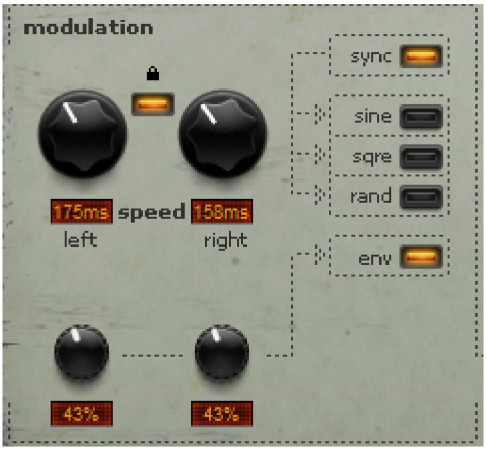
The actual filter circuit provides the following parameters:

- Cut-off frequency (**freq**)
- Resonance (**reso**)
- Internal saturation (**drive**)

The filter can be set up separately or together with the link button.

You can set up the cut-off frequency or the resonance for each channel as modulation targets. Just like with CORVEX and ECOX, the small knobs on the lower row specify the modulation level.

For the modulation source, in general the same applies as with the previously mentioned effects. In addition, FILTOX provides the modulation of the filter section via an envelope follower. In this case, the input signal itself serves as the modulator. This makes it possible to produce well-known "auto wah" sounds.



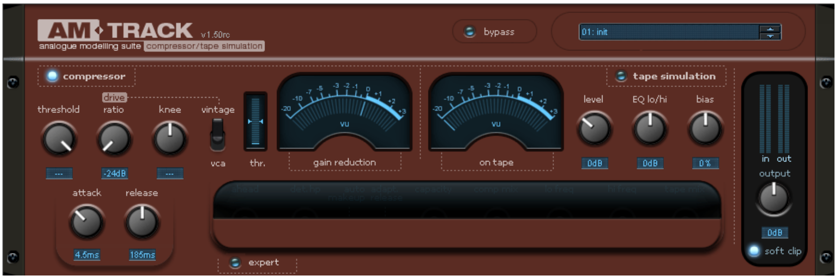
To use this envelope follower mode, proceed as follows:

- In the modulation section, press the "**env**" button.
- The two small "**gain**" controllers on the bottom help customize the input level for setting the envelope. The LEDs at the top no longer show the LFO speed, but rather indicate the set sensitivity visually.
- Usually, the sensitivity is controlled so that signal peaks activate the LEDs more brightly. If gain is set too low, you'd have to set the knobs unnecessarily high for the modulation targets. If input levels are too high, the behavior will not be noticeable. Additionally, the problem of overmodulation arises when the detector circuitry of the envelope follower leads to inaccurate tracking as a result of saturation. This tracking is also influenced by the following parameters:
- "**Speed left & right**" controller: These set the LFO speed. In "Envelope" mode, these may be used to set the attack & release of the envelope for each channel. This means that minimum speed settings can cause a quick increase of the envelope to produce quick modulation access. Since attack & release are coupled settings, quick attacks correspond to short release values. This way, the control voltage for the filter drops more quickly than at middle or slower settings.

- You should adapt the speed as precisely as possible to the signal. Times that are too short can cause errors in tracking and fluctuations, while times that are too long will miss short signal peaks. Internally, however, the detector works semi-automatically for release time, so that the setting is less critical than with a purely manual method.
- The lock symbol in the modulation section has a special meaning in envelope mode: once active, both channels are linked together for detection so that panning effects may be based on different controller settings, and not because of a stereophonic input signal.

Analogue Modelling Suite: AM-Track

(Samplitude Pro X6 Suite)



AM-Track Analogue Modelling Compressor & Tape Simulation

"AM-Track" is a combination of an analog compressor and a tape simulator in a single device. The plug-in was developed for specifically editing and enhancing input signals (instruments, vocals). The audio signal may be enhanced and made more lively with a combination of compression and tape saturation in order to assert itself in the entire mix.

An explanation of the compressor section of the AM-Track, its special advantages compared to "conventional" software compressors, and the available parameters follows. We will also explain the features of tape simulation.

Compressor Section

Two completely different compressors work in AM-Track, each with their own independent control and sound methods.

You may be wondering why we mention sound when talking about a compressor, since compressors merely relate to control actions. This isn't as simple as the idea of “making loud quiet.”

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

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

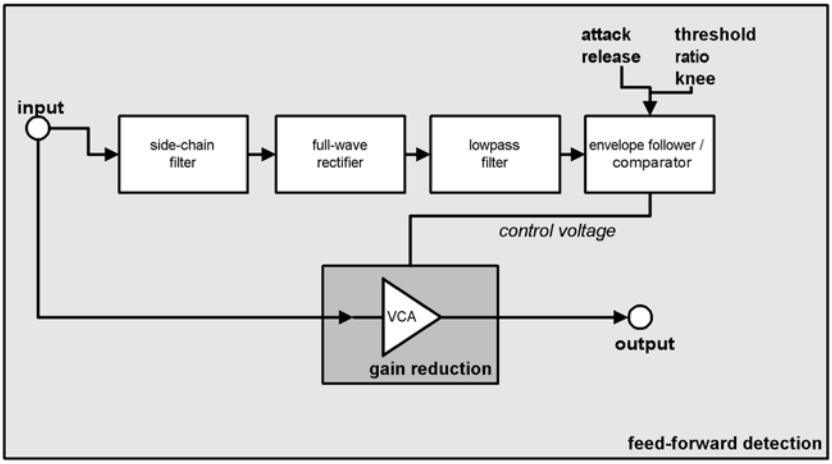
VCA Mode

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

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

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

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

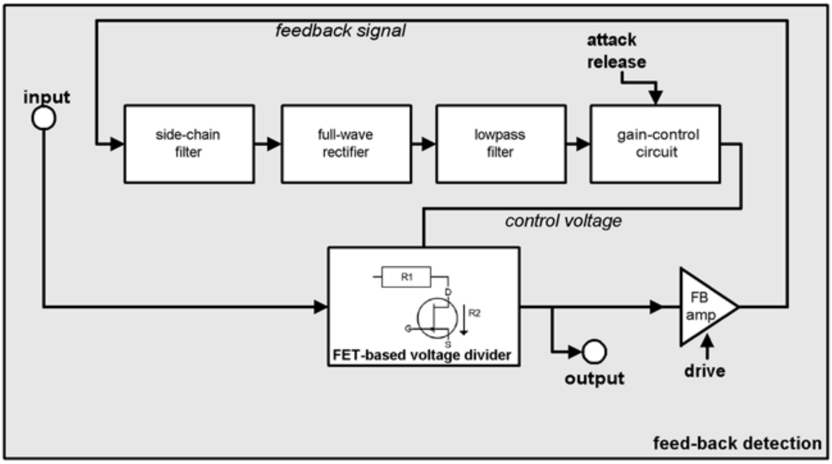


In contrast to this, there is a “feedback” method which provides a second compressor mode.

VINTAGE mode

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

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



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

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

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

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

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

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

Compression Parameters

VCA Mode

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

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

Note on attack & release: In general, short attacks are used for moderate compression and making the transient response softer; longer times retain the "bite" of a specific instrument at larger compression rates or make the sound a bit snappier. With more difficult sources, like a very dynamic vocal track (ballads), for example, you can use a longer attack so that the project runs more smoothly and quietly; the release time may be trimmed audibly to match the pauses or the song speed.

Shorter release time may be used for modern, aggressive "close up" vocals, e.g. when breathing sounds are an important stylistic device and the voice should sound very full and compact.

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

Vintage Mode

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

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

Compression Expert Settings

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

- **Look ahead**: AM-Track is always ahead of the signal. You can specify how many milliseconds you want to "look ahead." The audio signal path is delayed according to the signal route so that the detection circuit is fed first with the input signal (so-called "Look-ahead delay"). You can increase the attack time and still avoid fast peaks. The latency compensation in the host program ensures that other tracks in the project are adjusted and that no time delay occurs. For percussive signals, you can even set the delay all the way to “0.”

- **Detector hp filter**: This high-cut filter is positioned before the two compressors' detection circuit. You can use it to specifically exclude basses and mids from these rules. Complex signals with bass and hi information like a subgroup or complete mixdown produce fewer “Pumping” artifacts. This is because low-frequency signals feature the most power and therefore always trigger regulation and modulate other frequency ranges in the volume
- **Auto makeup gain**: Normally, you have to continuously adjust level reduction to generate “compression” at the same maximum level. This is done by activating auto makeup gain. The volume difference expected from the set working parameters is determined and applied as an output factor after master regulation. If you prefer to adjust the “classic” level reduction and amplification manually, you can deactivate this function.
- **Adaptive release**: This is “semi-automatic,” i.e. you can roughly adjust the release time, and AM-Track reduces it according to the current signal power from “a little (1%)” to “Considerably slower (100%).” In “Vintage” mode, this regulation method is particularly intense, since it affects the feedback loop process. For instance, if you are editing vocal tracks or dense, complex material, it can sound “calmer” or more “musical” if adaptive release is activated.
- **Capacity**: Adjusting the “capacity” controller sets the time response of the “adaptive release.” The greater the capacity, the more sluggish the release adjustment. You can therefore influence larger parts of the compensation response. For instance, if you want to use vocals that have been “moved forward,” you should use a short release time (maybe 80-100 ms) and a greater value for semi-automatic (e.g. 80). Vice versa, you can reduce automatic feed by switching the relation (smaller capacity, generally greater release time).
- **Comp mix**: Parallel compression is a popular “studio trick,” particularly with complex material. Adding the original signal retains the transients and spectral balance of the source. You can add compression by turning the mix controller. A mixed signal is particularly discreet, more transparent, and less “squishy” with vocals, whereby the compressed portion usually has a higher level reduction than without adding the original.

Tape Section

The tape simulation in AM-Track comes after the compression section and offers you the opportunity to give your recordings an "analog touch" by reproducing typical aspects of a tape recording.

What also happens in this case is that the magnetic storage space of the tape becomes exhausted and the signal more distorted when the recording level is increased. There are other factors as well however, e.g. pre and de-emphasis. Since saving on tape happens depending on the frequency, pre-emphasis makes sure the dynamic area is used to its full extent, according to a normed characteristic curve (e.g. NAB, EBU). This is because the signal cannot simply be transferred unfiltered at high levels onto the media via the recording head. The pre-filter creates a characteristic harmonic spectrum for possibly intended overmodulation by the user, which then changes according to the level of saturation.

With the de-emphasis circuit, the pre-filter may be undone with an inverse characteristics curve on playback. However, distortion caused by tape saturation might shift the spectral balance.

As a result of the enormous complexity of the process of magnetic recording, additional factors also influence the acoustic result, like, e.g. pre-magnetization (bias). This relates to high-frequency voltage swing (usually in sine form at 150 - 200 kHz) which is then applied to the tape via the erase head before the same part passes by the recording head a few centimeters later. This erasure current ensures the equal orientation of the magnetic particles and keeps the hysterisis loop intact, which is important for the magnetization process, and functions qualitatively. Low currents do have a brighter sound picture as a result, but magnetization is inefficient and the maximum recording level is therefore quite low. On the other hand, a bias current which is too high is associated with losses in highs, but does allow for higher levels with less distortion.

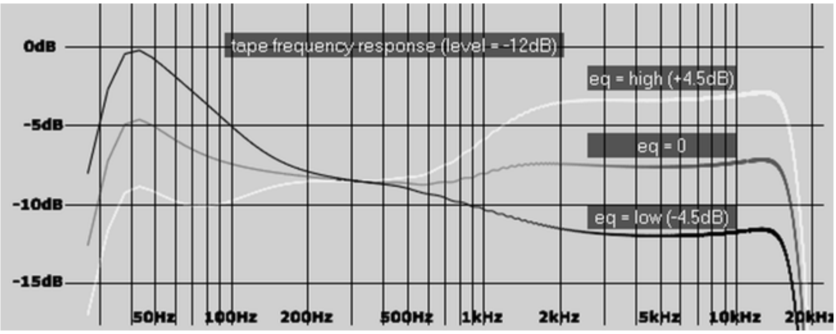
In addition, memory effects when recording tapes are responsible for a part of the characteristic sound, since time-dependent factors play a role in the feeding of the tape reel along the heads, e.g. mutual inductance and self-erasure.

Regarding the simulation of these processes, we have concentrated on some aspects of the "real world" and have created a virtual "machine" in AM-Track, which permits the following interventions by using the controllers on the interface:

- **Level**: Sets the input level. This determines when the "tape" is saturated and how strong the effect of the coloring/soiling is. You simultaneously gain more loudness.
- **EQ low/hi**: Adjusts the frequency response (spectral balance controller). You can choose whether you would like to have an output signal of high bass level or whether it should have more highs. This alters the pre-emphasis at the "recording end" as well as playback equalization. In "Expert" mode, you can vary the employed frequencies for lows and highs for distortion and equalization. However, note that the frequency response for the simulation will not be neutral,

even if the "EQ Low/Hi" controller is neutral. There will always be some slight frequency-selective amplification.

- **Bias**: Moving the operation point (bias adjust). Increasing the bias will result in a higher "recording level", but tape saturation will occur sooner. You will also increase the above mentioned loss effects, and the result will be a (dynamic) reduction of the highs. Turning the bias controller to the left results in the opposite effect: the highs are not reduced, but the signal level stays lower.
- **Tape mix** (in the "expert" section): Anything that applies to a parallel compression can also be applied to the tape section. Non-edited transients in particular are responsible for perceiving "speed", "liveliness" and "freedom", but may be lost in case of too much saturation/overdrive. Mixing in the original slightly enables you regulate the band quite far while still retaining the so-called attributes.



Tips for Applying Tape Simulation

The harmonics generated during saturation can quickly cause a state of "acoustic fatigue" for the listener, particularly for high-frequency rich material and/or a frequency shift at the benefit of the highs. A direct 1:1 comparison with the tape section switched off reveals slight differences. Subtle settings are generally sufficient for complex signals in order to create a slightly "analog touch".

Use the tape simulation as a "peak stop" if you use one of the two compressors to compress the signals: Transients that the compressor allows through (e.g. when using longer attack times) may be gently blocked by "taping" the signal afterwards.

AM-Track also contains a soft clip circuit at the output to block overmodulation. The audio result is certainly not the same as tape saturation. In practice, the soft clip function stays switched on to provide an emergency brake, especially if the plug-in is in the last station of the master slot. The soft clipper limits the signal with a soft characteristic curve at -0.1 dBFS. It's possible to switch off the function if you can predict good level responses or if the following stages in the mixer/master chain will not have a problem with higher levels.

Analogue Modelling Suite: AM-Pulse

(Samplitude Pro X6 Suite)

AM-Pulse is a "transient modeler", a creative tool for editing envelope and sustain processes (attack and sustain) on percussive or dynamic signals.

In particular, the input of the signal is an important factor in the acoustic perception. The so-called transients are extremely short oscillations in the make-up of a sound which enable the human ear to precisely differentiate sounds from one another and localize where they come from.

Alongside transient modeling this effect also allows you to enhance the signal or even distort it. For this to work in the am-pulse, individual steps as "virtual hardware" have been set up and designed using elements of models and components "borrowed" from the analogue world in order to achieve smooth, organic operation with character.

Possible instances where AM-Pulse can be used:

- Drums (e. g. kick, snare, toms, subgroup signal): Increasing or decreasing the attack leads to a harsher or softer sound respectively, while increasing or decreasing the sustain changes the room information with overheads or ambience tracks.
- Acoustic or electric guitars and basses: Accentuate or dampen the attack, e. g. for playing with a plectrum
- Balance volume fluctuations and explosive sounds in vocal recordings
- Reducing background noises

Working Method of the Transient Section

At first glance, it seems as if AM-Pulse has a dynamic compressor working "beneath the hood". However, this is not the case. In comparison to conventional compressors, AM-Pulse is level-independent. For example, if you reduce the attack rate of a snare drum by 6 dB, this is independent of how loud it was played. However, you have to determine a threshold as if for a compressor. No editing takes place below this threshold.

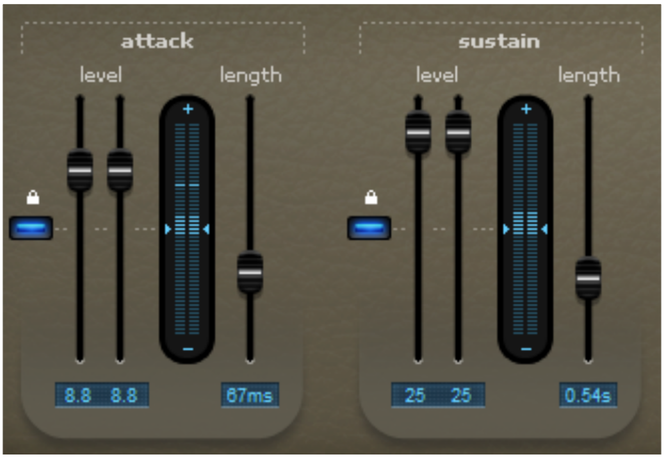
The possibilities mentioned for using AM-Pulse are based on the principle of "transient recognition".

AM-Pulse continuously analyses the input signal by means of so-called "envelope followers" that sample the time-based signal. At the attack and sustain phases, there are multiple envelope followers working which possess different attack and drop-out times. Attack and sustain can be safely recognized by means of continuous comparison measurements.

Internally, the AM-Pulse uses virtual VCAs ("voltage controlled amplifiers") that generate a control voltage from the resulting envelopes. You can use the fader to apply its value to the attack and decay rate. This voltage may be applied via sliders. This multiplies the input signal via the control voltage (controller in positive direction) or divides the signal by the voltage (negative direction.

Based on the principle of transient detection, the dynamic signals described above are especially well-suited for editing. The more percussive the audio signal, the cleaner and more predictable the control.

Transient Section Parameters



There are three faders on the front plate for processing attack and sustain:

- **Level** (left channel): An increase takes place above the line; moving the fader in the opposite direction reduces the level. The signal remains unaffected in the centre position.
- **Level** (right channel): As above. Normally, both channel faders are coupled together by default (as are the internal "detectors"). You may edit the channels

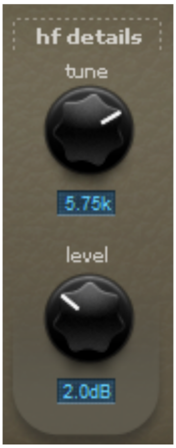
separately by depressing the button with the padlock symbol. The control voltages for both channels are then determined separately.
- **Length**: This determines how long the signal is scanned and maintained in the respective section. A lower value results only in a short amplification or reduction which sounds quite electronic. Longer times mostly sound more homogenous.

Note: High "length" values in the "attack" phase are especially in danger of being affected too strongly. The more complex the initial material, the more carefully "length" should be adjusted.

Saturation and HF Details Section

As a creative tool, AM-Pulse offers an adjustable saturation level and a "high-frequency details" module.

Both sections are located within the circuit after transient processing. Modeled sounds can thereby be processed additionally. There are many ways to combine these three sections to create unique sounds. We will describe a few of these scenarios below.



For **hf details**, exciter circuitry may be used to add artificial harmonics to signals at the set frequency.

The **saturation** section works similar to a tube pre-amp. This features the typical "tube-like" volume compression behavior and "drives" the signal into saturation at high levels. The "saturation" controller mainly determines the input gain of the tube stage. When turned fully to the left, no saturation occurs and this section remains inactive.

For the amplification of the signal, odd and even harmonics are created. The output signal first becomes louder and "richer". It becomes rougher and "dirtier" as the amplification increases, and finally sounds drastically distorted at an appropriately high input level.

The saturation of the signal is frequency-selective in this case: how much has to be pre and post filtered depends on the setting of the "saturation" fader. The more "saturation" the higher the filtering and sound modification. The internal degree of distortion is dependent on the setting of the level fader.

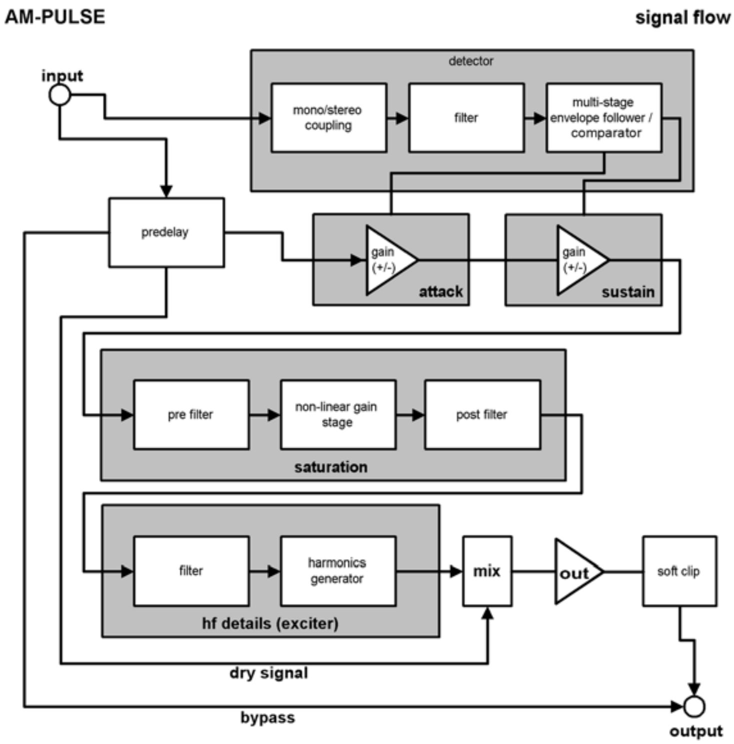
Use the **mix** fader to set how much of the processed signal is in the entire signal.

Creative Tips for Working with AM-Pulse

We have added a few presets to AM-Pulse for selection from the preset menu of the console. Several of these are ideal starting points for the following experiments:

- Attack & sustain are often contrary processes. For example, if you would like to add more "bite" to electronic drums by accentuating the attack, experiment with reducing the sustain level. Both sections then often require less "drive" and the sound generally has far calmer effects than only one of the two sections, but at double the level.
- To increase loudness, sounds with a lot of attack but a low average level may only be "tamed" with the saturation level. A few "saturation" dBs are usually sufficient.
- Try to simulate several amplifier types: combine the transients and saturation section. With a slight reduction in attack (about 2-3 dB) and a slight saturation (maybe 6 dB), you can achieve results featuring soft signal edges, slightly increased loudness, and a bit more "life" in the signal. You could possibly add 1 - 2 dB "HF details". The result would then come very close to a tube amplifier.
- If you would prefer slightly more filtering from the saturation section (the above-mentioned "loudness") but less "crunch", move the "input" fader down slightly and increase the "saturation" controller correspondingly. Keep an eye on the peak meter in this case, too. Subtle changes already have a large influence on the overall impulse behavior of the signal and slight colorations can even be heard in the A/B comparison.
- Set the attack and sustain of the left and right channels to opposite settings so that the signal "moves" in the stereo field according to its own transient data. This "auto panning" function even works with mono sources and is ideal for creating new and unique loops.
- Drastic changes can be smoothed with the mix fader. Among other things, entirely new sounds can also be created. The result usually sounds different than a simply softened setting of the fader at 100% mix. For example, you can use sustain with acoustic drums to break out of the room, but keep the integrity of the recording with targeted mixing of the original.

Much like AM-Track, AM-Pulse contains a soft clip circuit at the output for making sure overmodulation doesn't occur. The method mentioned with AM-Track also applies here.



Analogue Modelling Suite: AM-Phibia

(Samplitude Pro X6 Suite)

AM-phibia is a tube amplifier/channel strip. It combines an optical compressor with a pre and post filter unit. The filter presets offer appropriate settings for various types of input signal. While interacting with the compressor section AM-Phibia can be used as a vocal pre-amp, tube guitar amp or simply for creating a warm sound when mixing or mastering.

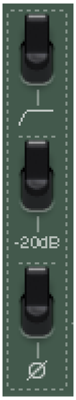
Typical applications of AM-Phibia:

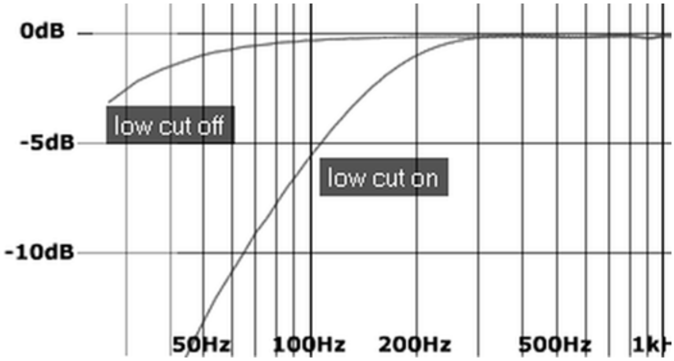
- As an optical compressor (the classic LDR-based light sources are modeled)
- As a valve pre-amp
- As a sound pre-processor with corresponding EQ and compressor settings
- As a guitar tube amp modeler with pre-amp, sound processor and speaker cabinet simulation
- As a signal processor for warm sound synthesis with all selectable tube stages
- As an audio "mincer"
- As an audio refiner
- As a tube exciter

Many application options can be selected from a comprehensive preset list.

AM-Phibia - Signal Flow and Stages

AM-Phibia – 1) Reverse phase, input pad, low-cut



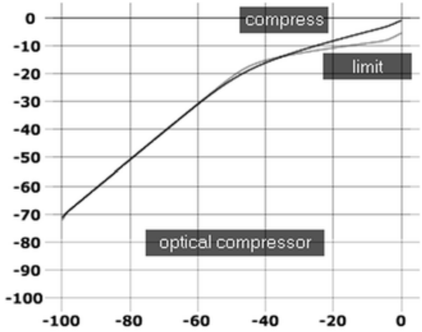


AM-Phibia – 2) optical compressor

The compressor of AM-Phibia is one of the most gentle compressors available on the market and is ideal for difficult signals such as vocals. The fundamental circuit is an emulation of a classic opto-coupling design featuring a so-called feedback circuit.

For this principle a light source (LED, light bulb, or electro-luminescent panel) is supplied the output signal. The emitted light is beamed onto a light-dependent resistor and changes to the resistance result in a change in the amplification factor of the entire circuit (controllable voltage divider). The light source is fed from the output instead of the input due to historical, practical reasons: the circuit gently oscillates to the signal and is very stable without additional settings, and operates very musically so that in hardware reality, this type of device can be built with a minimum amount of required components (which improves sound quality). Due to the overall lag of the opto-coupling system, optical compressors have a relatively sluggish release reaction. This is accompanied by the so-called memory effect of the opto-coupler.A longer exposure to light also means longer return times. Due to the feedback control, the control times and compression ratio also highly depend on the control times, and the compression ratio strongly depends on the input signal.

Specify whether you want to tap the feedback signal after the gain stage or the second filter. Frequency-dependent compression can also be achieved in combination with the pre-filter stage.



The compressor may also be switched between compression and limiter.

Note that the ratio of program regulation depends on the program.

AM-Phibia – 3) prefilter

Different types of EQ are available. Each is treated as an independently structured circuit.

- active A/B

This is an emulation of an analog filter module that operates via positive and negative feedback loops. In this case, A and B differ from one another in terms of the input frequencies of the bands. B is optimized for use with speech or singing.

- passive A/B

This circuit corresponds with the classic Baxandall network and is comparable to stereo systems and some guitar amps. Type A and type B differ slightly; the bass and highs are spaced out further spectrally for type A, and the mid range is more extensive for type B. Variant A is intended for general applications, filter B is optimized for vocals. The mid range, which is also available in this case, is not available in a Baxandall network and has been made possible in the existing circuitry as cascading low and high-pass sections, much like in common passive equalizers. The interesting thing about this setup is the effect the individual steps have on the entire phase response, which contributes to the unique sound of this circuitry.

- guitar passive

Classic switching such as on Marshall/Fender amps. These parameters are highly dependent on each other (more highs = less bass; mids are also influenced). Just like general "passive" switches, complex phases are also present in this case, a result of variable mixing of individual branches in the filter network and the typical charm of classical design.

- guitar active

A convertible partner for a typically "American" high gain sound. The parameters do not influence each other as much as the passive variant, although a slight amount of internal feedback is applied to give the circuit a bit more power. Of course, this also influences the phase response, which also provides its own character.

- bass passive

A full-tube Peavey bass pre-amp was the source of inspiration for this EQ (T. B. Raxx, similar to Alpha), with a circuit that resembles various Marshal/Fender styles. In this case however, the bass branch does not depend as strongly on the other settings, but a high level of interactivity is present. If the mids are "weakened", the sound turns quite "hollow", which is ideal for slap basses.

- bass active

Similar to "guitar active". In this case however, the corner frequencies are designed for bass guitars.

AM-Phibia – 4) gain

The heart of AM-Phibia is the gain level, an emulation of class A tube circuitry. Simply set the gain setting, and the "magic" happens inside; the frequency response changes, resulting in the right limit containing fewer highs.

A small amount of make-up gain is used to automatically compensate the relatively high amplification factor, and typical valve-like distortions and harmonics occur at the upper end without becoming too loud. Normally, high-gain circuit simulations feature a weak spot at this high a degree of non-linearity, i.e. so-called "aliasing". Harmonics created by distortions ideally exceed the audible range. At a sample rate of e.g. 44 kHz, unpleasant artifacts can occur due to signal mirroring of the Nyquist frequency (22.5 kHz) in the audible range (without harmonic relation). This is particularly annoying in the case of distorted guitar sounds, e.g. if you "bend" at the 24th fret when playing solo and aliasing becomes apparent via bending in the opposite direction. For this reason, we always leave the internal sample rate of the gain stage between 176 and 192 kHz to minimize artifacts.

AM-Phibia – 5) postfilter

This section only provides a low and high parameters; the cutoff frequency point of both may be adjusted, however (see "Expert view" below).

- active

The same parameters as the pre-filter stage, except that the mid band is missing.

- passive cut

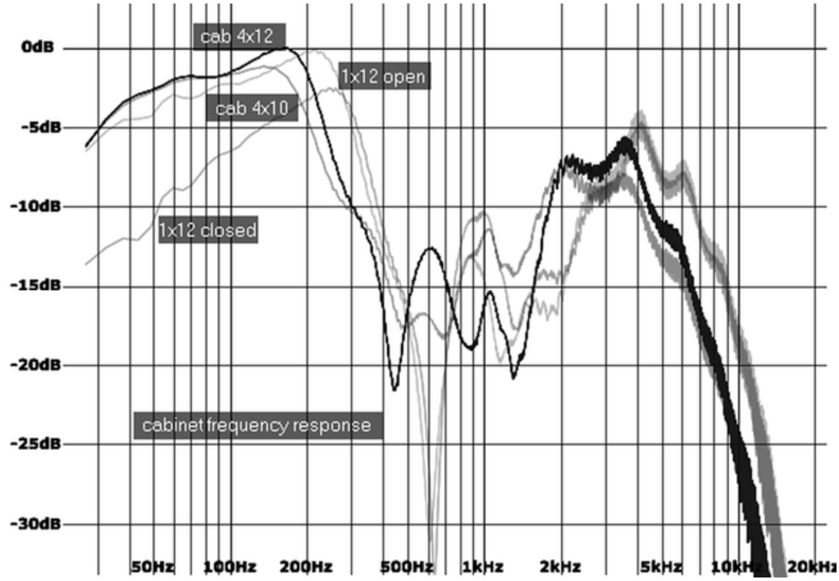
Baxandall (cut) network similar to the pre-filter, but without mids.

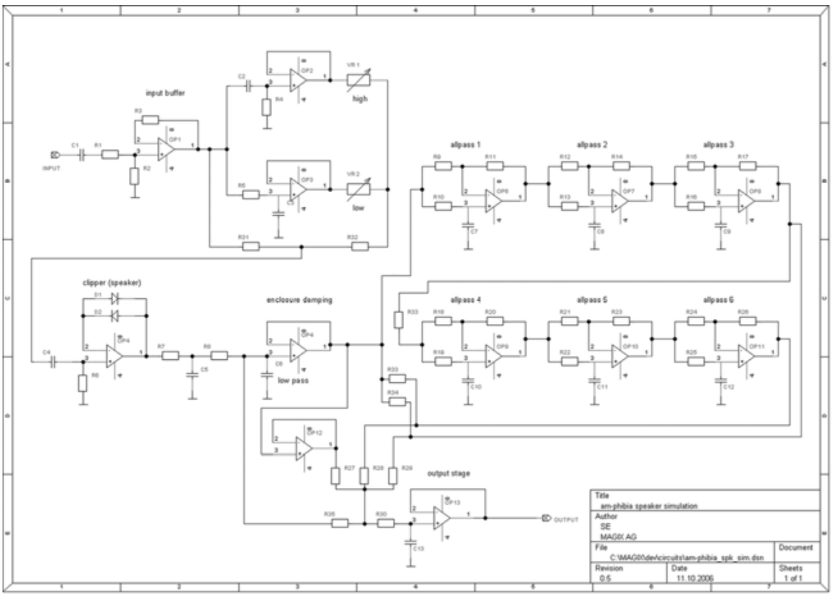
- exciter

Amplifies the low and high frequency ranges via frequency-dependent saturation. The control elements mix the saturated part with the original, which is common with traditional exciters. Due to the additional harmonics created during saturation, the exciter circuit can sound very different than a conventional EQ in case important spectral ranges are not present. The bass range may also seem fuller due to the switching applied. Despite the immense sound possibilities, use the exciter carefully, since the ear is fatigued by listening to this type of signal and due to the high energy density.

- cab simulations

These sound control networks are a departure from "official" EQ settings. The speaker cabinet simulations imitate the sound of classic speaker/case combinations familiar from conventional collections of guitarists and bass players. In line with the "analog by design theory" of AM-Phibia, we've also focused on switch-heavy interaction in this case. AM-Phibia works with an armada of non-linear amplifier stages, filter circuits for frequency-response changes, and complex phase-shifting processes. To begin with, we examined the sound of the speaker and how it becomes distorted depending on the levels, and we also studied its characteristic frequency responses in relation to its environment.This speaker emits direct sound from the front, while the back of the cone swings in the opposite direction so that the waves enter the cabinet phase-inverted. More sound-altering characteristics occur here: the sound is directed at the enclosure, partially absorbed (e.g. by insulation material), partially reflected, and mixed with the direct sound. Resonance effects such as bass reflex openings, static waves, return effects on the speaker, and so on, are also apparent here. We also added properties of typical models to our speakers such as bass reflex tubes in the signal transit time, which achieves a "typical" sound that would not be achievable with impulse responses alone. An important aspect in this case is the distortion created by the speaker at higher levels.





AM-Phibia – 6) volume

Volume is compensated using a valve circuit that adds atmosphere. This circuit ensures that the output signal does not exceed 0 dB.

AM-Phibia - Expert View

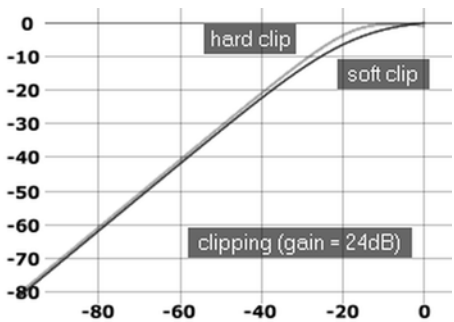
Opening this view mode gives you even more opportunities for editing. Some decisive sound parameters of the device are accessible directly from here:

- opto mem

This changes how the optical coupling device addresses the compressor and thus the related memory effect. You can use the "opto-mem" controller to work directly with the parameters for the system latency and influence the degree of program dependency. Transients generally have less influence on controlling than longer signals. If set to minimum, the circuit will recover relatively fast. If set higher, the release time also increases according to the duration of a loud signal.

- clipping

This function controls the behavior of all tubes in AM-Phibia. Soft clipping (0) generates soft sound characteristics in overdrive. Hard clipping (100) creates a more global volume, although it may be high-pitched if handled incorrectly. If this setting is used with guitars, the sound of the entire device may change drastically, particularly in combination with distortion sounds.



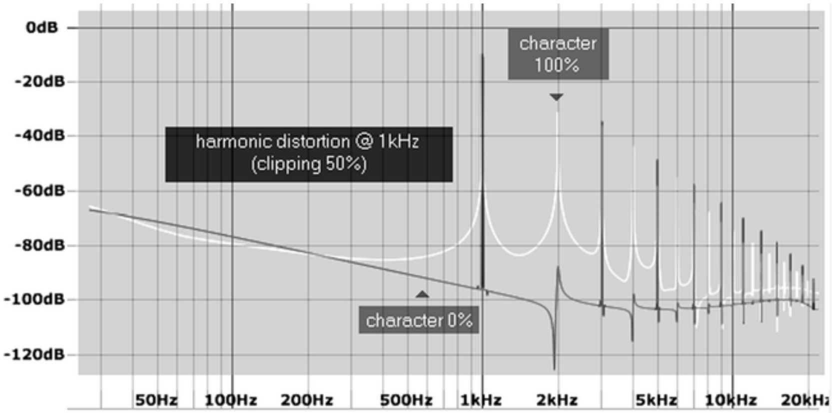
Guitar players can set this function correctly by using the volume controller of the guitar to control the distortion.

You will thereby most likely find the optimal setting at the point where AM-Phibia harmonizes best with the input signal.

With different sound sources, these experiments are less difficult and are worth trying out.

- character

This controller works together with clipping and the gain button. Sometimes, however, this effect is very subtle, meaning that an immediate effect is not audible with all effects. If AM-Phibia is used as a guitar amp simulator, the controller can determine most of the sound. "Character" only works on the tube bias (control voltage for regulating the electron flow within a tube). Capacity changes are also influenced by this reaction, since tubes feature a memory effect. This way you can observe a slight bass increase and more constant harmonies.



A higher "character" value with loud and high bass signals can easily add "oomph", although there is a risk of the sound being rougher in lower ranges. Discreet use of moderate gain settings can sound similar to "transformer-linked" devices.

- pre MF freq

This controller allows you to change the mid frequency of the pre-filter section. The available range is dependent on the underlying model

- post LF/HF

These controls change the corner frequencies of all controls of the post-filter section (basses and highs).

Analogue Modelling Suite: AM-Munition

(Samplitude Pro X6 Suite)

AM-Munition: compressor/limiter program

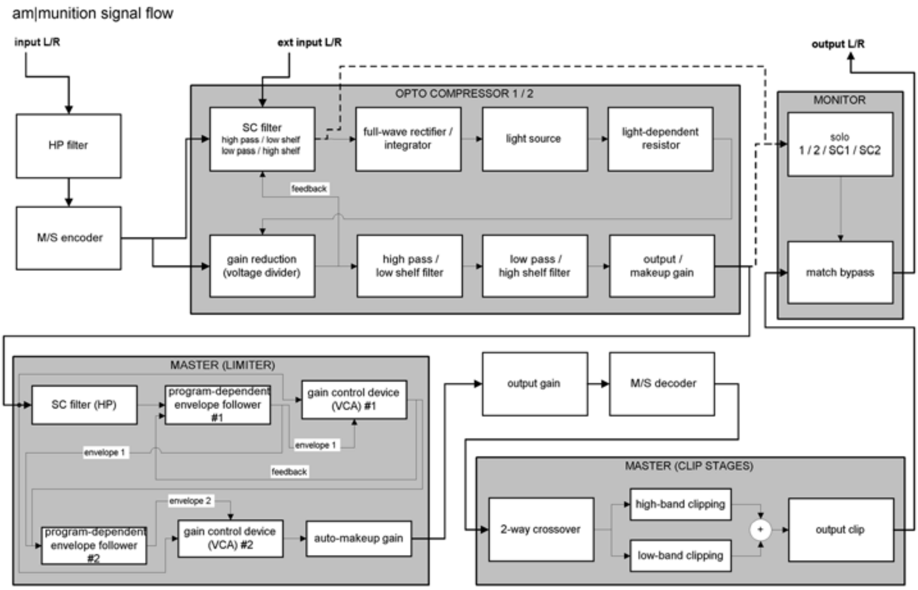
AM-Munition is a compressor/limiter for spicing up the mix and giving it more kick. This mastering and dynamics tool makes the mix not only louder and adds definition, but can also be used as a limiter. In the process, transients are conserved and a compact, but nonetheless loud signal is delivered. This occurs via a relatively slow serial compression and a soft clipping for the stray level peaks; AM-Munition is sidechain-capable and can also be used for M/S processing.

The **AM-Munition** plug-in is an extremely versatile, dynamic tool for editing groups or signal sums, especially in the area of mastering. It has separate units like compression, filtering, sidechain, limiter, and clipper. All modules and parameters are optimized to perform their fundamental functions without any compromises: effective enrichment of the program material without causing bothersome artifacts, a high maximum volume, and an "analog" behavior with an individual sound signature. This follows the maxim of allowing volume without distorting important transients. This is especially a problem with excessive use of the popular brickwall limiter. In fact, many digital limiters drastically increase the average level, but only at the cost of vitality of the sound, usually resulting in a fatigued audio effect. In drastic cases, snare hits are perceived as noise or hissing.

AM-Munition attempts to maintain the attack during increased compression of material. In this manner, the plug-in also functions as a trick: the limiter level is slower than it would be otherwise. Those transients which occur will be blocked in a multi-step soft clipping unit upon output. This clipper works frequency-oriented to enable a relatively high signal level already "solo", without sounding unpleasantly distorted. With the proper settings the combination of limiting and clipping results in a livelier sound with more punch and character.

AM-Munition appears inconvenient and complex compared to other traditional dynamics tools at first. However, with its combination of available steps, modes, and parameters, the plug-in offers creative flexibility and provides a powerful tool in the hands of ambitious sound engineers.

Overview of AM-Munition



AM-Munition is composed of the following sections:

Mode: The basic functionality of the plug-in is specified here. The normal mode is "stereo", meaning the input signal is entered on 2 channels (left and right) to the sidechain and thereby to the control circuit. In M/S mode, however, encoding the stereo signal will be carried out in parts as "center" (left and right) and "side" (left-right, differential signal). After this, both parts are sent to the sidechain. The functionality of M/S mode is exceptionally well suited to compression of material from the pop and dance genres, where bass sources are commonly positioned in the center of the stereo field. Normal compression in stereo mode will always produce modulation or "pumping", since the bass drum or the bass line reduces the total volume with its own energy. However, the division of the signal into center and side information merely makes a stronger modulation in a different range of the stereo field more noticeable.

OPTICAL COMPRESSOR: The heart of AM-Munition is made up of two compressor units which are indicated by "1" and "2". Depending on the selected mode, you can edit the left and right or the center and side signals. The compressors are based upon a very detail-oriented modeling of an optoelectrical circuit. The principle in this case is very simple: the input signal (Side Chain) is used to control a light source. The simulated light hits a photo resistor (LDR, light dependent resistor), which produces changes in resistance depending upon changes in brightness. These in turn directly affect the input resistance of the compressor switching. One could say that the signal source is its own volume potentiometer. In this case, the important factors are system capacity (light source, its controls, and the applied photo resistance) and memory effects.Amazingly, circuits based on this principle sound very musical and open, and they are well suited for the compression of program material, provided the actions are not supposed to happen too quickly.

Besides controlling the light source to achieve a specific reference line and envelope curve, AM-Munition compressors enable the ratio of the input signal to edited signal to be mixed. This parallel compression offers far-reaching possibilities for the subtle compression of signals. Because a part of the control circuit is bypassed, important transients of the signal spectral information are maintained. The compressed portion can often be output in depth simultaneously.

Both optical compressor units offer the possibility to restrict the frequency response to a particular output. In this case, a combination of high and low pass as well as shelving filters come into play. For example, filtering is useful for editing the side signals of basses and lower mids in M/S mode (keyword: vinyl cutting) or for restricting the cymbals of the percussion to the edge of the stereo field.

SIDECHAIN: As mentioned earlier, the sidechain signal is responsible for what the compressor sees. The sidechain value is usually a smoothed mid value of the input or a peak. By editing the frequency response of this signal, the relationship of the entire compressor circuit changes rapidly. For example, basses which are filtered out in the sidechain can maintain controllers against unwanted pump artifacts, and in the ideal case this also increases the volume in bass heavy pieces.

A mix of output signal at the plug-in and an external source offers additional possibilities for sound production, especially during a mastering session when single groups are being edited.

MASTER: The master section features one of the most important circuits of the plug-in for sound compression and volume enhancement: the limiter. As is the case with the input, this is combined at the output with a soft clip stage. This makes a fast reaction time or even a look-ahead unnecessary. At the same time, a skilled choice of limiter thresholds produces an airy, natural, fundamental sound, even under difficult conditions and high level reduction. Similar to the sidechain filter of the regular compressors, the limiter also provides the possibility to remove low frequency signals from the controllers.

The limiter switch in AM-Munition works in two stages (see switch image). The first stage has the primary task of extracting an envelope from the input signal. This is transferred to the second stage in place of the actual signal, which in turn forms its own envelope. Both stages work with different parameters and provide different perceptions of the signal. Furthermore, the first step may be switched between forwards and reverse detection. This cooperation between the stages makes a very exact temporal image of the amplification process possible, especially in difficult situations featuring bass-rich signals, which should normally be compensated for, but produce disturbing inter-modulation noise with regular limiters.

The limiter circuit connects the clipping stage, which is composed of two serial units. The first forms a dual-band clipper together with a 2-way frequency-separating filter. The advantage of separated clipping of basses and highs (mids, too) is a higher achievable loudness with fewer perceivable disturbances in comparison to broadband disturbances. This sort of disturbing intermodulation and roughening of the complete sound image may be reduced, especially with clearly-defined bass material.

The applied frequency filter is modeled on the Linkwitz-Riley filter (4th order, 24dB/Octave, Butterworth characteristic), which is often found in speaker construction. This produces an adequately sharp separation of bands and ensures an exact phase length as the bands are combined at the output. The complete phase shift of the 2-band clipping stage amounts to 360 degrees. This should be monitored when AM-Munition is used parallel to other tracks.

Clipping at this stage and the broadband which follows at the output has an adjustable intensity (see "Config" page below).

The second clipping stage (output end) provides the output's safeguard, and this may be used "solo" for creative intervention.

The clipping at both stages may be set without stages from hard to soft overmodulation. This makes a complete series of "analog" sounding combinations possible.

MONITOR section: The individual compressors (1 and 2) or those of the sidechain may be set to the sum for selective monitoring ("solo"). This makes it possible to adjust the level of signals being edited so that in active "match" state the volume complies with the volume of the unedited bypass signal. This will produce a psycho-acoustic phenomenon: louder signals sound better and can be realistically and easily checked to see if the quality of signal being edited is still in tact apart from volume increase.

Metering: The center of this plug-in interface features several level displays. Among those are two VUs, two peak meters, and two stereo meters.

Both of the outer VU meters display the effects of the main compressors when set to the "reduction" position. The centrally placed displays visualize the value of the reduction caused by the master limiter. When set to "output", all four level displays show the output; the VU meters provide the present RMS value (root/mean/square, average level), while the peak displays measure peaks.

A 0 dB reference value may be set for the RMS value. This is always useful if you would like to have a clear view of the average level. For example, to produce a song or an album with an RMS of -11 dBFS (a rather conservative value when one considers current chart productions, but lower headroom always has a destructive effect). In this case, you would set the "RMS 0 dB ref" controller to "-11 dbFS". If you place the metering switch of the output level to "output" for monitoring, the VU meters will show the displacement of -11 dB. If you read -4, then the RMS level is actually at -15 dB.

Both of the stereo meters are very useful for determining the range of the finally decoded stereo image and for comparing it to the unedited input signal in M/S mode.

AM-Munition Parameters

MODE

M/S/stereo: Selects functionality; 2-channel stereo or middle/side encoded.

Link: At 0%, both portions of the signal will be treated discreetly by the sidechain (L/R or M/S). 100% complies with that of a mono summation of the detector signal. Middle and high values are sensible for complex signals, which sometimes show deflection to the side of the stereo field.

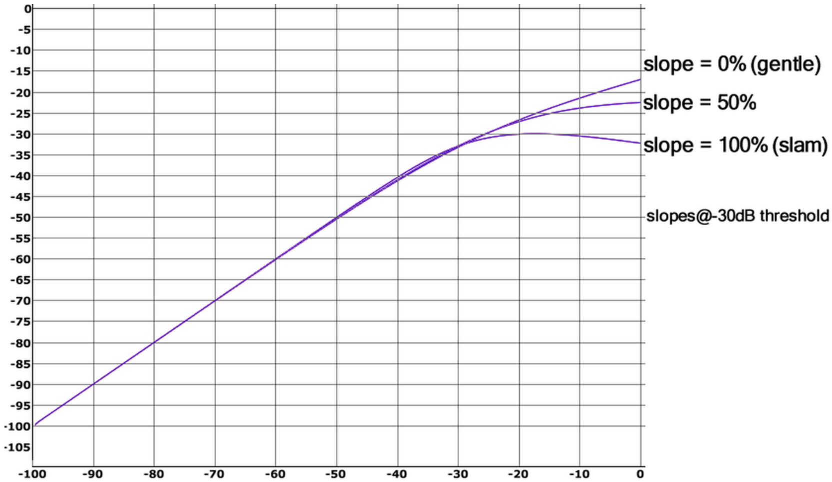
The settings of this controller affect compressor 1 and 2 and the limiter in the master section.

OPTICAL COMPRESSOR

Threshold: This controller defines the threshold at which compression occurs. From a technical standpoint, you control (in contrast to the fader) the signal strength used to activate the virtual light source with this value. The exact trigger point of the threshold is dependent upon the selected compression reference point and internal parameters. For more exact control of this threshold, the VU meters can be set to "reduction".

Output: This parameter is useful for gain increase at high level reduction. We recommend setting the output so that the resulting level matches that of the disabled compressor level.

Slope: This parameter is like the "ratio" when compared to a regular compressor. This is not, however, completely correct. In this case, the compression reference point is influenced so that the result is a softer reference point for lower values and higher values cause limiting or over-compression. Set at maximum, the reference point of lower threshold/higher input level even goes a little lower, instead of remaining at a limiter plateau. This is one of AM-Munition's specialties and is well suited for compression of percussion subgroups, e.g. when room features are supposed to stand out.



The slope controller does not change the reference point itself, but rather the way that transients are processed. Set to the minimum position, the compressors will operate in "feedback" mode, i.e. the input of the detector switch will be fed from the compressor output. This controller works fairly softly and transparently and gauges the signal gently, since existing deflections from the envelope are also included. The resulting reference line is therefore always quite mild.

The more the slope fader is set to the right, the more portions of the signal will be drawn into the compressor input. In this way, the controller combines "feed-forward" and "feedback". The leads to a higher weighting of the transients to produce an aggressive reference line.

We generally recommend a slope setting in the lower third of the fader for a less conspicuous transparent compression.

Response: The switching principle employed here provides musically sensible control times via selection of the "light source" and "photo resistance" system components. The combination of these components determines the shortest possible tune-in and tune-out times. With the response controller, the transient time may be extended to a wide range, since it's as if the image processing time is extended. A variation on this parameter will certainly have an influence on the complete functionality of the photo resistor, especially on "memory" effects. In this case, the current energy level of the program material decides the actual transient time of the compressor.

Comp mix: Input responsive parallel compression is possible with this controller. This is best adjusted to produce the highest sound transparency together with high level reduction. Any possible compression artifacts are skillfully masked to bypass portions of the signal, and important signal peaks and tuning-in processes of instruments can occur unhindered.

Filter (high-pass/low-pass or shelving filter): The filter switch on the output of the compressor unit sets a higher or lower frequency spectrum limit on the signal. For instance, this is useful in "M/S" mode, for filtering basses in the side signal, which otherwise unnecessarily reduce signal volume and would even lead to problems with phasing of stereo sources during vinyl mastering.

Both filters have gated characteristics (high or low-pass) on their outer ends, and these change by altering the cut-off frequency increasingly in a shelving curve with maximum 12 dB reduction. This results in relatively soft filtering that is ideal for basic corrections to the spectral balance.

Controls (key symbol): Connects corresponding controllers of both compressors and is useful in "Stereo" mode. Separation in "M/S" mode should be deactivated when settings on the filters or output controller are adjusted, e.g. to filter out bass signals from the side signal or to set the width of the bass

SIDECHAIN

Level: This controller allows fine adjustments to be made to the external signal. An adjacent level may be monitored with the peak meter.

Source: Adjusts the mix ratio between the program and an external signal. This mix forms the sidechain signal evaluated by the compression circuit.

> **Note**: To enable external sidechaining, please ensure that "Sidechain input" in the menu bar of the plug-in window is activated and that the relevant source is routed to this additional bus (stereo).

Sidechain filter: Describes the functionality of both controllers for the filter switch in relation to the filter network in the compressors. By limiting this the frequency field, the ranges may be excluded from detection or weighted less.

MASTER

Output: AM-Munition's final output level. Make sure that the soft-clipping switch is located at the final stage of the output. The function is to block signal peaks passed on by the limiter.

We mentioned at the beginning that the philosophy of AM-Munition's sound signature is based in part on the division of work between limiters and clippers. For high signal volumes, the qualitative results are dependent on threshold and output values. A high signal level at the output will inevitably be bound with a high portion of clipped signal peaks. As the case may be, please use the "clip" display to monitor this. This displays the difference between clipped and unclipped signals. The higher the display level, the more clipping results.

Threshold: This parameter defines the threshold for the limiter on the one hand, and also increases the output level after successful level reduction ("auto-makeup gain"). This corresponds with the conventional operation of brickwall limiters, since only a few parameters are needed for quick, targeted loudness.

Release: Sets the return time of the limiter until the controller achieves the output point after falling below the threshold. Note that some kind of program automation monitors the signal independent of the set value. For this reason, you will not find any exact controller times (e.g. milliseconds). The value entered could be understood roughly as the circuit direction. Lower values provide a very fast control response time and weight the program automation quite lightly, while higher values produce increased interaction between both of the limiter's detector stages and participating envelopes.

Sidechain filter (high-pass symbol): For difficult tasks during the compression of material, for instance with pop or dance productions, it's often useful to reduce bass segments or remove them completely. In this way, pumping produced by the limiters can be avoided. The sidechain filter of the limiter can be increased in two steps for this. In this case, always ensure that the bass range is not subject to limiting like the remaining frequency range; the clipping stage needs to intervene and limit the signal as required.

MONITOR

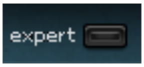
Solo: By pressing buttons 1 or 2, one or both of the compressor outputs will be sent to the monitor line. In this way, you can bypass the master section on the one hand, and watch over the effects of the compressors on the other. This can be very important when working in M/S mode.

The sidechain may be monitored via "SC1" or "SC2" as required.

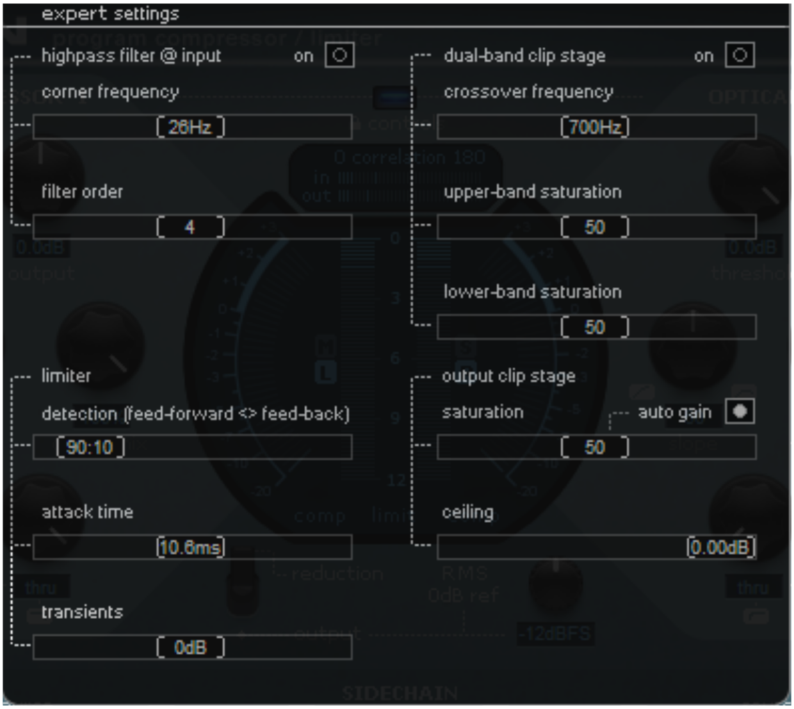
Match bypass: A tool for increasing volume is useful for audio projects, since things usually sound better when they are louder. In order to properly determine whether your material has retained its own quality after these processing stages, it is important to carry out a volume comparison with the unedited bypass portion. The "match" button is provided for this purpose. This works together with the corresponding controller as a signal dampener.

Since the match function lies at the final position of the signal path, you can obviously also use it to remove audio material with a lower level than that defined by the output clipper as 0 dbFS via the plug-in.

AM-Munition CONFIG



Click the **expert** button to access additional AM-Munition parameters.



The regular compressor view will be hidden and a separate display will appear. This alternative view shows a series of additional **preset-based options**:

Highpass filter @ input: At the plug-in's input, there is a high-pass filter with variable cut-off frequency and edge steepness. Especially for mastering applications, this can take over the tasks of "cleaning up" the signal before additional processing and filtering out low-frequency sections which aren't required or could possibly cause overmodulation/reduction of the signal's maximum loudness.

The filter applied features Butterworth characteristics (q=0.707) and may be switched from 2nd order to 4th, 6th, or 8th order.

For exact filter settings, we recommend working with subwoofers in good spatial-acoustic conditions.

The following options involve **fine settings in the master section**:

Limiter

Detection (feed forward <> feed back: Limiter switching may be set without stages, whether the signal envelopes are to be gained directly from the input signal (forward), or whether they are drawn from existing amplitude process generated control voltage (feed back). The latter approach often creates a calmer, more balanced sound image and weights transients higher than slower signal sections. The resulting threshold of the limiter will be simultaneously rounded off a little more (soft-knee). This usually causes a slightly reduced total volume. Generally, the sound image in "forward mode" is a little more direct and gripping. Sometimes, however, the effects are quite subtle, and in some circumstances, these are only perceivable when levels are reduced substantially.

Attack time: Normally, a limiter should feature an extremely fast response time. For "look ahead" circuits that work with delay in the signal path, the attack time may be relatively long versus the delay, supplying the limiter with enough room to work correctly. With a "zero-latency" design like ours, on the other hand, it basically has to be assumed that the limiter does not get any time at all to adjust. Theoretically, it has to be infinitely fast.

AM-Munition utilizes the cooperation of limiting and clipping. The clipper blocks that at the output which the limiter lets through to signal's transient response. Sometimes, the attack time can be accordingly generous. Lower values of less than 5 ms are available if small level changes need to be smoothed out. Larger values, for instance percussion, may be placed at the forefront of the total signal in contrast to complete instruments or groups. For example, with an adequately long attack time, a snare can maintain its punch in a complex signal and the complete sound can be kept loud and compact. Conventional applications include harder rock or metal productions (a clipper at the output is a better partner than a brickwall limiter anyway, since a limiter tends to consume important transients). The combination applied in AM-Munition, on the other hand, is able to transform amplitude energy into spectral energy.

Dual-band clip stage

Crossover frequency: This parameter defines the separation frequency at which the signal separates into a bass or higher range.

Upper-band/lower-band saturation: When this controller is set to the minimum setting, a hard clipping sound will result as full overdrive is achieved. A high degree of additional overtones/harmonics will appear at the clipping threshold. Below this level, the sound image will remain completely neutral.

Larger values of this parameter produce softer clipping or saturation of the signal. Initially, this highlights only even numbered, lower-order harmonics (mainly so-called "k3" harmonics). For example, a sine pattern of 1 kHz is added to an overtone of 3 kHz and lower amplitude. At 100% saturation, the overtone spectrum is quite similar to that of analog reel-to-reel machines, if driven via saturation of the tape as with full overdrive at the limit.

Output clip stage

This also enables the balance between hard clipping and softer saturation to be fine tuned.

Auto gain: This function increases the total level prior to the output by 3 dB, depending on the clipping settings. This is influenced by the softer saturation characteristic, the maximum of which is normally already reached prior to complete overdrive, although in this case, this value is raised by this amount to provide the maximum amount of headroom. Without "auto gain", loudness will increase as a result of the changed overtone spectrum if the absolute level remains the same.

If "auto gain" is active, this level increase will be avoided and consistent loudness is achieved independent of the saturation curve. In this case, the input of the clipper is lowered by the expected level gain at the output to reproduce a "consistent ratio". This is especially useful for the dual-band clipper, since duplicate saturation of the signal without damping may sound overdriven.

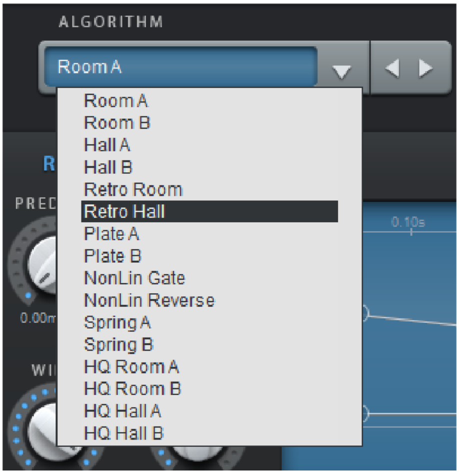
The "auto gain" function therefore ensures a certain degree of room in case additional overtones are generated.

VariVerb II

VariVerb II (short for: VARIable reVERB) is a high-quality reverb plug-in that runs on an algorithmic basis. It provides a series of rooms, halls, reverb plates, spring reverbs and so-called non-linear effects, which can be edited comprehensively and easily.

"Algorithmic" refers to the reverb impression not being created using impulse responses, as is often the case nowadays. Instead, the calculation is based on mathematic models just like those in classic reverb devices. With VariVerb II these algorithms can be chosen from a list, each specialized for a specific replication or a specific purpose and given its own set of parameters. In contrast, technology based on convolution is only able to interpret a static copy of a process, i.e. it doesn't know the room. In fact, realistic reverberation is associated with dynamic, non-linear, high-grade, interactive processes. Not only do these conditions make sure that the artificial reverb method is constantly updated, it also underlines the effect it has as a tool of artistic expression. Let's call this audio aesthetics.

VariVerb II – algorithms

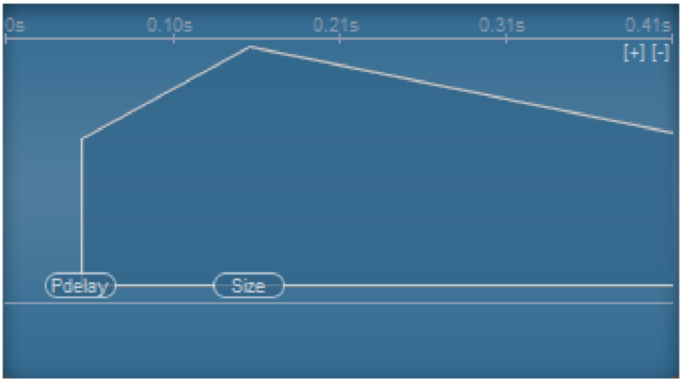


The algorithms in VariVerb II pro specialized in a specific type of "room creation" can be selected from the list on display:

- **Room A:** Small to medium-sized room, dampening of the low frequencies, immediate response, high density/diffusion
- **Room B:** Room similar to A, less dampening, slightly different reflection pattern, responding a bit slower
- **Reverb A:** Larger hall, slower reverb buildup, less density than the "rooms"
- **Reverb B:** Similar to reverb A but later response of the reflection, slightly "colder" sound
- **Retro room:** Classic room effect in the style of well-known hardware reverb devices from the 80s/90s. More artificial algorithms than the previous methods. The vintage/retro character takes center stage here, definitely an "effect" sound.
- **Retro hall:** Type of creation like retro room, but this is a classic hall effect with the corresponding reflection pattern. Thick, cloud-like reverberation tail.
- **Plate A:** Classic reverb plate, high diffusion, quite "dark" and "heavy", slight panorama effects
- **Plate B:** Reverb plate similar to A but with lighter character, faster response
- **NonLin Gate:** Non-linear reverb, no sustain but abrupt end instead (classic "gated" reverb effect)
- **NonLin Reverse:** Reverb with reversed amplitude, sound impression played such as "backwards"
- **Spring A:** Model of a reverb spiral with two linked spring systems, typical echo effects and excitation noise
- **Spring B:** Similar to spring A; however, the vibration is "softer"
- **HQ room A/B & hall A/B:** Extremely high quality and very complex simulation, which, however, is associated with high processor strain. Nevertheless, or maybe because of it, it's the definitive main hall algorithm. Very plastic, tangible room impression with free positioning of the sound source and the listening location. Here, the A and B models differ slightly from one another like in the regular room and hall algorithms, i. e. HQ room A provides somewhat different dimensions and different composition than B. With the HQ halls, hall B communicates with A later and sounds slightly more complex.

The individual models and available parameters are described further below.

VariVerb II - overview of details



The **Display** is the core of VariVerb II, visualizing the available parameters, their current values, and graphically presenting the reverberated envelope.

The preset list is located above the device interface in the **console**, where you will find the typical settings for various everyday studio work tasks.

For general AUX routing there is a separate category featuring all models available in a standard setting. This section can be used to design reverb (if you want to start from scratch). The mixing ratio is preset to 100%.

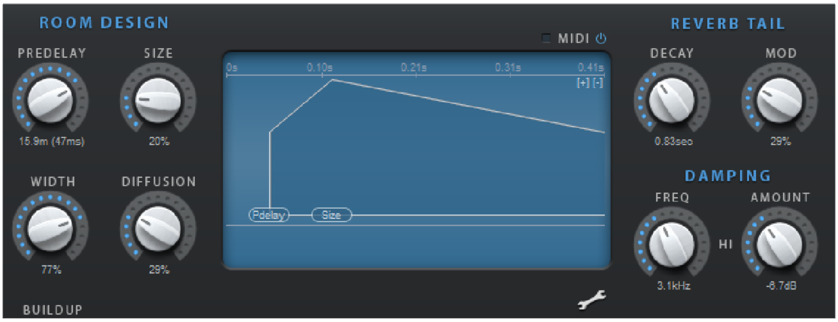
DRY/WET: The controller for the mix ratio is located on the right of the main interface (direct signal and reverb portion).

EQ LO / HI: Allows you to "pre-filter" the signal before adding reverb. The filters have shelving characteristics. The trigger frequencies vary according to the reverb algorithm.

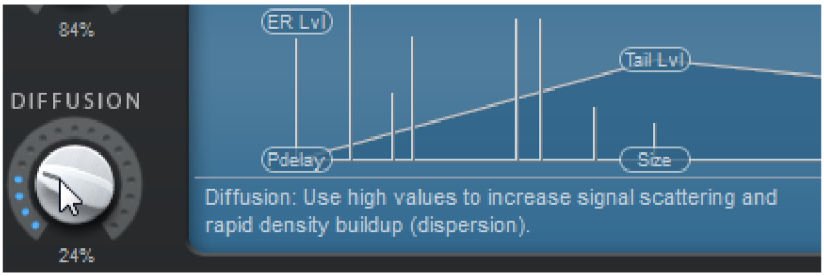
Further to the left you will find the "**Mute**" switch.

You can use these to temporarily switch off the input signal during longer reverb decay periods, i.e. if you want to judge the resonating reverb precisely.

Next to the display you'll find controllers for reverb parameters such as room size and decay.



Parameters will appear as a "tooltip" on the display if you hover the mouse pointer over a controller.



The following documentation focuses on available algorithmic models and individual parameters. Some apply to several models, in which case they are only referred to once.

Room & Hall

The four available room models (room A/B & hall A/B) are quite complex. Unlike with the regular digital process, with VariVerb II the reverberation is not created by simple echo summation. Such simple systems do not account for the fact that the echo structure becomes denser as the sustain time in real rooms increases. VariVerb II uses a network of reflections that resemble natural processes.

Here you will find the following options:

ROOM DESIGN:

- **PREDELAY:** Determines the time by which the actual reverb process is delayed. This parameter is ideal for fine tuning in order to convey a feeling of "distance" or "isolation".
- **SIZE:** Determines the room size. Moving the slider to the leftmost position means the smallest size, moving it to the right extends the reflection times. Smaller "Size" settings also reduce the distance between the individual reflections. Resonance may also develop due to the short temporal distance between them. Increasing the size of the room creates more space, but this also means that the so-called "modal density" decreases, i.e. more time will pass before the impression of a confined room develops.
- **WIDTH:** Specifies the width of the stereo effect. If set to the left, the reverb is monaural.
- **DIFFUSION:** With this parameter you can simulate diffusing at irregular walls and objects. The higher the value is set, the softer and denser will the reverb appear.
- **ABSORPTION:** VariVerb II calculates 16 initial reflections (compiled to 8 in the display). This controller lets you specify how far the last reflections are weighted. A lower value results in a more "lively" sound impression, although this may also have a more intrusive or unsettled effect. A higher degree of absorption means a smaller amplitude as well as damping of highs.
- **ER:TAIL:** The mixing ratio of the first reflections (early reflections) and subsequent reverberation. These reverberations are decisive for perceiving the room size. Mixing them into the actual reverb improves the possibility of locating voices or instruments. Missing reflections at the start often result in "spongy" sound characteristics without orientation in "room" and "depth".

REVERB TAIL:

- **DECAY**: The reverberation time. With this controller you can define how far the echo will be absorbed, that is, the time for the reverb to die away. You will then only hear the first reflection. Turning the knob to the right minimizes damping and results in long, sustained reverberation.
- **MOD:** This parameter regulates modulation, which introduces a slow variation of delay times.

DAMPING:

- **FREQ HI:** The frequency at which highs will be dampened.
- **AMOUNT HI:** Allows you to influence the frequency-dependent absorption of the reverb. A higher value dampens the highs and upper mids similar to the absorption effect of the air and in particular the material composition of walls.

Plate

A real plate reverb consists of a large metal plate (often 0.5 to 1m² thick or more), that is put into motion by a magnet and coil system (similar to a loudspeaker). The plate is normally hung on springs so that it is able to oscillate freely. Various types of damping are possible to reduce the reverb time. The reverb plate features so-called "taps" at various locations. These are sound pickups similar to those on a guitar. Several of these taps are combined to make up a full signal. Reverb plates are usually mono, i.e. a stereo signal is added to the plate as a sum. A (pseudo-)stereophonic signal is created by combining taps and their position on the plate.

Real reverb plates are only rarely used nowadays and have been almost entirely replaced by software simulation. They continue to be highly popular due to the very dense sound (high diffusion) and inaudible discreet echoes. They are therefore ideal for percussive material. For vocals, plate reverb generates a flattering luxurious effect. The slightly "metallic" resonance of a plate can also be used to generate a vintage effect.

Parameters:

- **PREDELAY**: See room/reverb
- **SIZE**: Size of the virtual reverb plate
- **WIDTH**: See room/reverb
- **DIFFUSION:** See room/reverb
- **BUILDUP:** Low values result in a fast reverb increase. If the value is greater, the activation phase increases
- **DECAY**: Reverberation time, see room/reverb
- **MOD:** See room/reverb
- **FREQ HI:** See room/reverb
- **AMOUNT HI:** See room/reverb

Retro room / Retro hall

As already mentioned, in addition to realistic generators, VariVerb II also features vintage-like algorithms used by many well-known manufacturers to this day. The used processes already have the highest-possible "dispersion" at the start of the reverb phase and generate a dense, wide sound. This is anything but realistic, since discreet reflections are mainly perceived as flat echoes. This sound is usually ideal for breakthrough power and foreground effects. In the past, various manufacturers have attempted to tackle very brief repeat loops and static patterns resulting from insufficient memory. An interference occurs in this case that also adds charm to this algorithm.

Parameters:

- **PREDELAY**: See room/reverb
- **SIZE**: Size of the virtual reverb plate
- **WIDTH**: See room/reverb
- **DIFFUSION:** See room/reverb
- **ER:Tail:** See room/reverb
- **DECAY**: Reverberation time, see room/reverb
- **MOD:** See room/reverb
- **FREQ HI:** See room/reverb

- **AMOUNT HI:** See room/reverb

NonLin Gate / NonLin Reverse

These (non-linear) models are the only ones without a real counterpart in VariVerb Pro. Non-linear reverb is based on a sequence of individual delay blocks (so-called bursts). By weighting these blocks, you can create envelopes. Two of these typical envelopes are "nonlin gate" and "nonlin reverse". These may be set as follows:

- **PREDELAY**: See room/reverb
- **SIZE:** The size of the individual "bursts" i. e. the length of the total reverb summed up.
- **WIDTH**: See room/reverb
- **DIFFUSION:** Smoothens the reverb effect. The greater the value, the more echo blocks are blurred and the more reverb occurs.

Please note that no decay parameter exists for these models, i.e. there is no "decay".

Spring

You may remember reverb spirals from guitar and keyboard amplifiers. At the bottom of these amps is a unit consisting of two to four spirals mounted on a vibration-free carriage. Like the reverb plate, it uses systems for transforming the electric signal into a mechanical one and the other way around. There are different designs and sizes of spring reverb; however, they all have the same typical sound: the splash-like "bloing" caused by movement of the springs. When the reverb dies off, the basic pitch of the spring(s) can usually be heard quite clearly. Furthermore, the frequency range is considerably limited due to the losses in the spirals and used pick-up/transmitter. Nevertheless, or maybe precisely for this reason, their sound is highly unique. Some music styles such as dub & reggae would hardly be possible without reverb.

The spring reverb is implemented in VariVerb II as a digital effect based on so-called "physical modeling" algorithms. Based on a mass-spring system, the effect operates similar to the physical model of a swinging string.

Parameters:

- **PREDELAY**: See room/reverb
- **SIZE**: Size of the spring system, i.e. length of the springs. A smaller value results in a very short response time and short echoes, greater values slightly smudge the typical oscillation sound of the spring. The length of echoes is increased.
- **WIDTH**: See room/reverb
- **SATURATION**: The spring models also include the processes when "outputting" the signal and "receiving" it via the magnet/coil system. Transfer of electric energy into mechanical power and vice versa is non-linear so that harmonic distortions may occur if the input level is high. The saturation parameter more or less increases the output volume and switches the system into saturation.

Depending on the material, a very interesting organic effect occurs which highlights the vintage character.

- **DECAY**: Reverberation time, see room/reverb
- **MOD:** See room/reverb
- **FREQ HI:** See room/reverb
- **AMOUNT HI:** See room/reverb

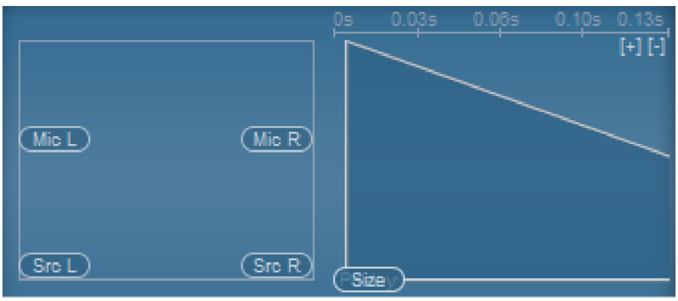
HQ (high-quality) models (Room A/B & Hall A/B)

The above-described room and reverb algorithms are of high quality, very versatile and require particularly low CPU power. If you want even more realistic and higher quality spaces and the load on your CPU isn't as important, take a look at the HQ modules. These simulate a room with rare plasticity and naturalness, almost as if they were based on impulse responses. But they are not; VariVerb II features an armada of delays and filter stages. These may require extra processing time, but they are still very useful.

In comparison to other modules, these algorithms cast a far closer meshed net over the virtual room. The results are natural, fast diffusion of the signal, high complexity of the reverb signal without echo pattern, and the possibility to freely position the two virtual microphones.

If the HQ rooms are opened in "Expert" mode, you will notice that editing for the first reflections is missing. The reason: in normal modules, the artistic and sound manipulating effect is in the foreground. A natural room, however, does not differentiate between early and late reflections. These migrate in time. The proportion of first reflections depends on the position of the audience.

In "HQ" mode, a reflection pattern and entire sound impression including stereo positioning may be edited via a top view of the selected room. The usual section shifts along with the reverb apron in the graphic display.



This includes freely positionable handles for sound sources ("Src L" and "Src R") as well as two microphones ("Mic L" and "Mic R"). For instance, if you increase the distance between the source and the microphone, you will hear the signal move away from you; as the delays increase, the sound characteristics become more "diffused" and overall more complex. At the same time, the previously harsher early reflections will disappear. This tool contains a lot of potential, and it may also be automated.

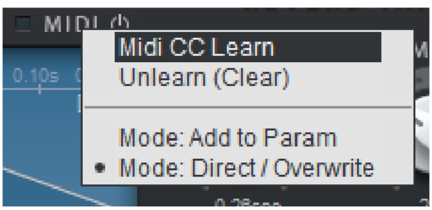
Plus, in "HQ" mode, you can also influence the dampening of highs as well as the reverberation time of the bass section. When interacting with the high band, realistic simulations are possible, e.g. wall surfaces. For instance, the wooden panels on the walls always reflect bass less than concrete walls (also due to possible gaps), which may also resonate and absorb the force of long-wave bass frequencies.

The same also applies to the furnishings of a room. Furniture, shelves, etc. are traps for bass.

HQ room/Hall model parameters:

- **PREDELAY**: See room/reverb
- **SIZE**: Size of the virtual reverb plate
- **WIDTH**: See room/reverb
- **DIFFUSION:** See room/reverb
- **DECAY**: Reverberation time, see room/reverb
- **MOD:** See room/reverb
- **FREQ HI:** See room/reverb
- **AMOUNT HI:** See room/reverb
- **FREQ LO:** Bass dampening frequency
- **AMOUNT LO:** Bass range dampening level.

MIDI CC Setup



If you activate one of the buttons in order to save, any following external controller movements will be used to generate the MIDI CC number. This enables source destination pairs to be saved one after the other.

If all parameters have been set, save the setup by pressing the "Save setup" button. If you leave the MIDI CC view before doing this, the previous settings will be restored.

This controller setup may be regarded as global, i. e. the settings are independent of the selected presets and the current project.

Vandal

(VANDAL in Samplitude Pro X6 Suite/ VANDAL SE in Samplitude Pro X6)

Virtual guitar & bass amplifier

Samplitude is a complete simulation suite for guitarists and bassists. This plug-in is capable of simulating the entire signal chain, from input to stomp boxes, amplifiers, microphoned loudspeakers and post-processing studio effects, all in top quality.

We focused on several aspects while designing this program, which make Samplitude a very special plug-in in terms of its amp modeling.

Version limits of Vandal SE versus Vandal:

1. The SE version of Vandal only includes the following stomp boxes (floor effects):

- La Crema Overdrive
- Hellfire Distortion
- Chorus
- PhaseShifter / Tremolo

2.MIDI scenes & MIDI controls are not included.

3.Guitar and bass amp, detailed microphone settings, and pre and power amps cannot be selected manually, rather only via "Presets & templates".

4. The SE version does not feature "Scene memory"; in this case, a preset only includes one individual scene. If a preset was created using the full version, then only the first scene will be used.

Special Features of Vandal

- High-fidelity analog modeling featuring accurate models of tube amplification stages, genuine real-world positive characteristics like distortion, dynamics, and audio complexity.
- Not the application of impulse responses during speaker simulation (conventional), but rather genuine physical modeling that calculates components like speaker, housing, recording space, and microphone individually based on physical situations, offering enormous freedom of sound design and realtime control.
- Easy-to-learn interface with clear workflows

- The focus is on the important components of a guitar or bass setup.
- Not a copy of a famous amplifier, but rather true "Custom Amplification Design" that features a sound of its own and versatility in terms of audio variety.
- Completely latency-free Samplitude generates no signal delay by doing away with impulse responses. Only your audio interface and driver determine the input and output latency during live playing.
- Full MIDI control of all parameters in each component.
- 4-times oversampling (below 44/48 kHz) for components that create distortion. This avoids typical digital artifacts (aliasing)
- Comparably low CPU load In a live setting your computer needs to be stable and have "room to stretch". In the studio, you may require multiple instances of the same plug-ins in one song. For example, for one specific metal riff you may require 10 guitars instead of two for that real "punch in the gut"; simply let Samplitude rise to the occasion in 10 separate instances...

VANDAL Quick Start

VANDAL Overview

We consciously designed the VANDAL interface to feature a realistic look. As bassists and guitarists ourselves, it seemed important to us to feel right "at home" here. The only unusual detail is that the floor pedals (and the gray carpet) are viewed from above, while the other effects are presented from the front. We admit, there was no way around this...

Also, no one in real life would place an insanely heavy amp on a 19-inch rack. But in this case, it seemed logical because the signal path flows just like how you are read these lines – from left to right, and from the top down.

The area at the top of the plug-in is called the "console". This is the so-called "switchboard" of VANDAL. The second half of this document (view page 945) will explore these areas in more detail.

Input level: VANDAL and the real world

Independent of the preferred software, amp modeling has to feature impeccable integration with audio hardware and ASIO drivers. Please ensure stable and latency-free operation and the largest possible signal-to-noise ratio of the converter or the complete analog section when selecting appropriate equipment. If in doubt, ask your audio equipment dealer for advice and consider your entire computer system. On-board sound systems are not recommended due to their rudimentary or unsuitable driver sets and the often below-average signal quality.

Before making any "serious" endeavors in VANDAL, please make sure that your audio interface is supplied with a sufficiently controllable signal level from your instrument. The input signal at the audio interface should be "saturated", but it should not be overmodulated. Compared to the possibly desired distortions created within the plug-in, an AD converter distortion will sound abrupt and horribly digital, and the the worst part is that you will never be able to remove it.

If very loud velocities sometimes reach the upper modulation limits and the average level stays at -12 or even -20 dB, everything is normal. The less noise your audio hardware has in the process, the better. We recommend working with devices that feature a resolution of at least 20 or 24 bits. A simple 16-bit sound card will stop being fun to use at the very latest when high-gain sounds come into play.

If possible, use an interface with a high-resistance instrument input (often labeled "HiZ"). Line or microphone inputs are not very well-suited for this; due to the relatively high impedance values of (passive) guitars or basses, this sort of input has a dampening effect on the instrument. The sound will seem dull and "dead" as a result...

If the interface does not feature an instrument input, insert a separate guitar pre-amp, a DI box, or even a simple pedal (not switched to "true bypass).

Instruments with active pickups or electronic components are less affected by this adjustment problem.

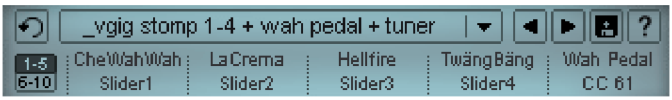
Please take this opportunity to also check the latency or the buffer size used by your audio system. A good ASIO driver should let you play with a latency of only a few milliseconds between the stroke and the calculated sound output from the software. Values under 5 ms are considered good. You may have to experiment to find the lowest limit at which your system functions stably. If you notice ugly crackling or even long drop-outs, then try a higher latency.

Input level reloaded: VANDAL's input level

After you have made sure that the external levels are clean, you should take a closer look at the input level; this controller is located at the top left of the console. Just like with a genuine guitar or bass setup, it's important to ensure the highest possible input level in order to be able to work optimally. This is even more important for distorted sounds and natural high-gain playing styles. You should make use of metering for this as well (the LED indicator next to the input control). If needed, activate the noise gate and adjust it so that it lightly suppresses the input signal during pauses in playing. VANDAL does not cut the input as hard as classic gates do, but instead regulates them finely via the signal energy, beginning with the highs, at the point where noise is most audible.

VANDAL preset and scene selection

Would you like to know about everything that VANDAL can do as quickly as possible? Just start strumming away and listen or flip through some of the included presets. These are available via the list in the upper edge of the console.



A preset includes all settings for the main elements of VANDAL, i.e. stomps, amp settings, cabinet simulation, studio effects, and a series of control parameters.

All settings for the main elements (such as the amp) are subdivided into "scenes" within a preset. This "scene memory" is located to the right of the preset list. You may create up to four variations of a preset using scenes. Numerous included presets make regular use of the scene concept. We will examine this working method (view page 946) more concretely later on.

Presets may be switched by clicking on the list or using the arrow symbols. You can also use the left and right cursor keys.

Scenes may be switched by clicking on the adjoining number (1 to 4) or on the name field of a scene. Alternatively, you can use the up/down arrow keys.

Of course, presets and scenes in VANDAL may also be controlled via MIDI (view page 949).

Presets are located in a subfolder within the VANDAL folder on the hard drive. These may be changed however you like, sounds may be exchanged with peers using VANDAL, and your preset library may be expanded without limits.

Components

Stomp Boxes (pedals)

The real world has produced a series of effect devices popular with guitarists and bassists in the "stomp box" format. We've also included a rich palette of these devices.

VANDAL includes four "stomp slots" for use with effects equipped from the list. The signal flow within this chain runs from left to right. Stomps can be moved around freely, by dragging an unused area on a pedal's surface.

When a stomp is assigned to a slot that's already occupied, the two change their places. Dragging while holding the [CTRL] key creates a copy of the pedal at the new location.

Each stomp box may be detached from the signal path using the "foot switch" or the blue button next to the selection list. An inactive device does not utilize the CPU.

Some stomps may be familiar to you; we reproduced some real prototypes with regard to functionality or sound, or simply got some inspiration from others. Some, in contrast, were created from the ground up and offer original sound ideas.

We're constantly developing the selection of existing stomps that are available. With each plug-in update, it is our intention to constantly supply you with fresh "food for thought" in addition to technical improvements. Look for updates to VANDAL at http://www.vandalamps.com.

The following stomps were available at the time this manual was printed (each in its own category):

Overdrive/Distortion

La Crema: Inspired by the world's most famous distortion pedal: the Ibanez Tube Screamer. La Crema offers soft musical overdrive sounds for solo parts, or simply serves to provide the amp with a prepped, contoured signal. The available gain is comparatively moderate; this is a typical pedal for classic rock and blues.

Halvar: The name may sound a bit clumsy and coarse. In reality, however, this blue Scandinavian is a real softie and is a homage to Yngwie J. Malmsteen and his preference for DOD distortion pedals. Halvar tidies up nicely in the bass range, and it's outstanding for shredding and styles of play where crystal-clear response without any bassy noise is important. The gain factor is comparable to La Crema, but Halvar distorts "harder" and has less accentuated mids.

Hellfire: Welcome to guitar hell! This pedal will roast everything, deliver gain until the cows come home, basses from the pit, cutting highs, and it easily creates the typical "scooped" sound of popular metal bands.

Fuzz: I can't get no Satisfaction, Foxy Lady. That basically says it all...

The Fuzz effect is famous for its singing-saw sound. Where overdrive and distortion effect the signal in a comparatively mild way to let it "breathe" within limits, Fuzz attempts to rebuild the sound from the ground up according to its own principles.

Bass Distortion: This pedal is of great service for bass players in search of juicy & meaty tones. For massive, chunky overdrive sounds, it is important to reduce low-frequency content from a distortion device. With classic fuzz tones, it's quite the contrary.

This distortion device delivers both, and many facets in between. It is equipped with a dual filter network, located before & after the distortion process. Simply tweak your favourite sound texture to taste.

Modulation effects

Chorus: The application and sound remind us of our personal favorites: those good old Boss pedals. Our Chorus offers modulation that's soft as butter but still crunchy and always assertive.

PhaseShifter/Tremolo: A phaser alone is already quite spacy, and a tremolo can sound a bit psychotic, but both together... Whether soft or choppy modulation (->Wave), slightly out of tune, or with sharp resonance effects... (-> Shift Q), it can sound as if it's up in the clouds or totally sick!

Flanger: Although technically related to the Chorus function, the Flanger uses shorter delay times and internal signal feedback to produce slightly edgier modulation effects. This Flanger pedal certainly sounds soft and wide in the same way that analog flanger pedals sound.

Bass Chorus: The essential sound of this pedal is very similar to the "normal" Chorus. Frequency response, delay times, and modulation curves are certainly optimized for use with bass. The Bass Chorus produces enormous width, but it never becomes muddy, no matter how deep it gets.

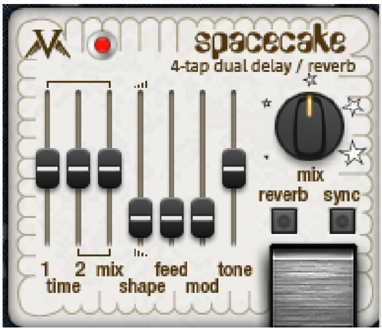
Delay / Reverb

Digital Delay: A simple delay pedal without too many frills that features maximum sound transparency. Simply set the time (via the delay controller), the feedback, and the damp level, mix it in, and that's it. As required, the delay time may be snapped in to the host software's currently active tempo via the sync switch.

TimeTunnel: Compared to the DigitalDelay, this pedal is a real vintage item. The (virtual) switching corresponds to that of the built-in "bucket brigade delay" in other stomps. Properties such as loss of highs in case of long delays or the "compander" switch for noise suppression of "breathing" sounds created by high feedback settings are clearly audible.
The BBD memory's virtual sampling rate may be modulated via the "Mod" controller similar to the Chorus effect, which enables very lively delay textures.

TwängBäng Spring Reverb: This is truly an authentic recreation of a spring-reverb system, incorporating all those lovely boingy, springy & slinky noises admired for decades. TwängBäng lets you select from a spring system using two seperate springs, or from one consisting of three springs. Both have similar delay times as of certain well-known US spring devices, but each one does have a distinct colour and reverb pattern on its own.

The 2-spring system sounds quite "vintage" while the 3-spring mode is a bit more balanced and open. Using the "tension" knob, the "springiness" and therefore the delay and attack structure can be altered. "Tone" & "damping" decide over the overall color and reverb time. Using a fair amount of „drive" makes the simulated transducers sweat a bit, creating a thick "glue" to the reverb sound.



SpaceCake Echo Reverb: This pedal is a perhaps unusual amalgam of a muti-tap delay and a reverb systems known from the late 1970s. Using proper BBD (bucket-brigade delay) emulation techniques borrowed from "TimeTunnel", the signal feeds two seperate delay lines here. Each delay has 4 taps on the output, which have user-adjustable weighting. This alone already lets you dial quite unusual reflection textures.

In "reverb" mode, things can go more wild. Now, the feedback signal of a delay line is derived from the sum of all 4 taps, and the two feedback paths are cross-coupled. Depending on the actual delay-time ratio of the two delay lines, very complex echoic and reverberant tails can be created. Just keep in mind that by summing the taps in "reverb" mode, the system is prone to instability. Luckily, you have the "mod" parameter at hand, slowly varying the delay times and keeping feedback under control. And it greatly enhances that "spaced-out" feeling.

Volume/Dynamics

Volume Pedal: No explanation needed. Step on it, give it some gas. You are, after all, driving with tube equipment! If the amp is set correctly, you can control distortion levels precisely just by varying the volume. Common swelling sounds are also no problem when using MIDI controls.

Compressor: The heart of this pedal is the modeling of a simple floor-board effect's typical control circuit, which is based on an FET (field effect transistor). FET compressors are famous for their speed, high degree of sustain, and above all for their character at comparably low switching complexity. Of course, the guitar signal may also be compressed subtly or only the attacks may be enhanced specifically.

Bass Compressor: The basic circuitry of this Stomp Box is comparable to the "normal" (guitar) compressor. Under the hood, adjusted specifications control at decisive positions to provide bassists with consistently solid sounds. The Bass Compressor works frequency selectively, i.e. very deep frequencies trigger the process less intensely than mids or highs. This enables the Drive controller to be used to produce a good foundation and thrust.

Twin-C: This compressor pedal divides the instrument's frequency range into two ranges (low & high) and sends these to separate compressor stages. This separation helps to avoid pumping artifacts or other typical problems caused by broadband compressors. The stomp selects control times automatically. Basically, very short attack times are applied to the upper band, while longer times are applied to the lower band. Twin-C enables slap bass sounds with extreme punch to be created easily, which offers a very sturdy foundation and a sound that "goes straight at you".

Tube Compressor: This stomp box uses virtual tube circuitry. Although in this case, it doesn't neccessarily mean the process is about distortion. It's more the fact that the gain-reduction process is a function of tube operation: like the "variable-mu" concept used in popular concepts from the 1950s onward, a tube can be externally controlled to change its amplification factor.

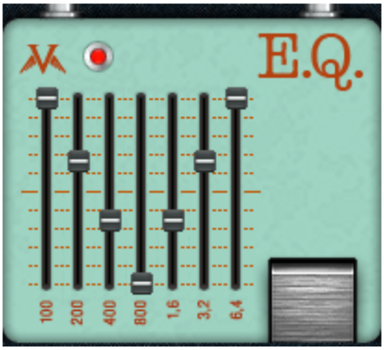
Of course, such a circuit can be driven hard by purpose and produce typical tube-like saturation textures.

In order to keep things well-balanced and controllable, a (symmetric) push-pull circuit is used in this design.

Using low to moderate "drive" settings, this pedal is a very smooth & gentle dynamics compressor.

Filter/EQ

CheWahWah: This little red rebel will prick up your ears. It can scream out your solos and fills loudly and clearly while keeping its cool and staying smooth. This is one of the few digital wah pedals that doesn't sound digital. On the normal setting, the pedal acts like a regular wah pedal; in auto mode, the filter circuit reacts directly to the incoming signal's amplitude. In this case, the pedal varies the sensitivity of the auto wah effect.



EQ: This pedal enables the guitar signal to be set precisely in seven frequency bands. This is not only suitable for corrective adjustments, it's also great for supplying any subsequent overdrive or distortion stages with perfectly prepared sound. It also performs well in solo passages, e.g. in case the mid range needs to be highlighted.

The heart of this EQ pedal is a circuit based on an analog model which features parallel filter stages. In this case, the typical disadvantages of serial filters such as phase problems and sound degradation are avoided. This EQ features the desired properties of real circuits, including added harmonics to the input or individual filter stages.

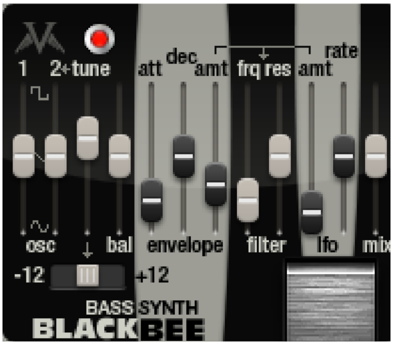
TrebleBooster: This effects pedal is ideal for that "British" sound or any kind of crunchy style. It effortlessly provides crystal clear highs to the amp or to other connected stages. Rhythm guitars come to life, while leads sparkle out of the speakers.

The underlying circuit remains unique in spite of a certain likeness to vintage treble boosters. The adjustable corner frequency ("freq" controller) is the key player: depending on the setting, dramatic changes to the frequency and phase responses may also be heard, which makes the booster extremely versatile.

FunkFrog: The ideal companion for sixties & funky styles, but also great for a lot more. In principle, this pedal is an auto wah; however, the frequency band is not only controlled by the input signal via an envelope follower ("env" controller). The "mix" controller allows an LFO (Low Frequency Oscillator) to be mixed in as the controlling signal. This creates a kind of rhythmically croaking auto wah, or maybe an oscillating, whirring filter thingy..

Others

Octaver: This effect uses classical frequency division switching to produce to sub-harmonics from the (mono) end signal. The controller "oct1" defines the portion of the first sub-octave (-12 semitones), and "oct2" defines the portion of the second one (-24 semitones). The direct portion of the signal can be mixed into this.
Both filter controllers enable the sharpness to be removed from the waveform that is produced, e.g. in order to thicken it up just for the bass range. As usual with octavers, clean tracking requires single tones; chords should not be played. In general, the selection of the neck pickup is also important, since this is where the basic sound is transferred most clearly.



BlackBee: This bass synthesizer can be used to manipulate the bass signal in a variety of ways. It offers the corresponding envelope and filter parameter plus two oscillators and an LFO. The "tune" controller can be used to adjust the pitch of the signal.

Amplifier

Vandal essentially offers two different amplifiers; a guitar amp and a bass amp. These can be selected from the list in the "Amplifier panel" below the stomp view.

As we mentioned, in Vandal we found our own way and avoided modeling specific amplifier brands and models. The amplifiers are set up variably so you to get a number of different sound characteristics out of your Samplitude amp. Internally, these circuit designs work exactly the same as the real versions. At a few points that are decisive for the final sound we've gone with our instincts and let our ears tell us what sounds right.

However, if you need a bit of help to get started with your sound you can go to "AMP + CAB TEMPLATE" and use one of the many presets.

Amp/Cabinet Template

The extremely variable concept for amps and cabinets within VANDAL is great for individualists, but it would take a bit of time to imitate a band mate's 40-year old Marshall with speakers of the same vintage. If you want to use a combination like this often, then it's nice to have a library for it, which is exactly what the templates are for.

Our team of experienced musicians has compiled a multitude of settings for the amp and speakers. These templates may be changed at any time, or you may create your own if a particular combination of amp and box is especially appealing.

Guitar Amp

Circuit Philosophy

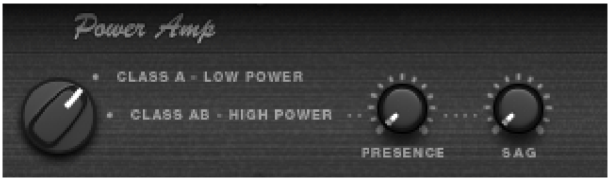
The VANDAL guitar amp offers three different pre-amp modes and two switchable end stage models. This basic configuration may be changed via the wrench symbol on the right side of the amp.

The pre-amp has three modes of operation:

- Classic (think Fender, early Marshalls, or the first Mesa/Boogie amps)
- British (oriented on Marshall Super Lead / "Plexi")
- Modern High Gain (design similar to "souped-up" amps like Rectifier, Soldano or Peavey 6505)

More about pre-amp modes and their properties resulting from channel selection can be found in the chapter "Pre-amp Channels" (view page 935).

The last stage has two modes:



- **Class A, low power:** A simple, low-performance, but musically pleasant distortion. In "A" mode, only a half-wave of the signal is amplified. To do this, the signal is brought to approximately half of the amplification range of the end tube(s) through bias voltage, because tubes always amplify only a half-wave of an alternating circuit. A class-A circuit may be put together using only a few components and sounds very "warm" because of its constantly asymmetrical characteristic curve shape (some even harmonics appear). This amplifier, however has the disadvantage of low efficiency or low output performance and a comparatively high liability of erosion. The latter has been left out of the modeling, of course...

- **Class A/B, high power:** In this case, each half-wave has its own tube, which (almost) doubles the efficiency. Class A/B router amps are more complicated to implement (and calibrate). Compared to Class A, the sound characteristics include descriptions such as "sovereign" and "powerful", but somewhat "colder", because they result in almost exclusively odd harmonics. Using positive and negative feedback via the output transformer, an additional "sound design" is often added here. For this reason, VANDAL's A/B power amp features more juice in its lows and more bite in its highs.
- **Presence:** On many amplifiers, this pot is located next to the EQ section, although a presence boost actually takes place inside the power amp circuit. Most push-pull amps use negative feedback (from the output transformer back to the power amp's input) to linearize the amplification process. Lowpass-filtering this feedback signal and mixing it (anti-phased) with the input results in boosting the mid and treble region. Using presence, the sound takes on a livelier and more up-front character.
- **SAG control:** Many older tube amps use rectifier tubes to transform AC current to DC (instead of conventional semiconductor diodes used today). However, a tube is a high-resistance component, and can't produce an even current flow during steep load changes. This "sagging" feature of a cranked-up tube amp is the acoustical result of these short-lived interruptions. Moderate sagging is initially perceived in the attack; it sounds somewhat compressed, but in a "lively" way. If the effect is even stronger, it changes the entire signal. Besides the dynamics, the harmonics spectrum also changes, because operating point of the tubes is shifted.
- Sagging occurs in this form only in push-pull end phases; in principle, Class A amps always draw constant (maximum) current from the mains.
- Intensive sagging will result in creating less "presence", as the overall signal gain in the power amplifier decreases, thereby generating only small voltage excursions to feed back on the power amp's input.

Preamp Channels

The guitar amp is set up with three channels in all preamp configurations (Classic, British, Modern High Gain).

- **Clean:** This channel should create no distortions. If the incoming channel is "hot enough", then it will be distorted in a typical tube fashion. The model will use only half of a double triode; in principle, a simple amplification takes place at the "working level".

- **Crunch:** Amplification stage number, circuit and, last but not least, the resulting sound all vary:
    - Two amplification stages are used in the "Classic" preamp setting. This model uses a long-established design that is similar to the classic Fender amps or the first Marshalls. The "classic" preamp delivers saturated lows that are only slightly dampened by the second stage. As a result, this circuit delivers the typical "brown" sounds of older vintage amps.
    - The "British" switch is inspired by dual input stages, such as Marshall "Plexi" variants (JTM, "Super Lead", etc.). The signal is sent to half of each triode circuit, and these favor different frequency ranges ("warm", "bright", etc). The "British" circuit in the VANDAL amp uses a set mixing ratio for both tube portions. In contrast to the "Classic" version, this preamp setting sounds more "alive" and reacts much more sensitively to dynamics, pick-up selection, and gain settings/guitar volume.
    - "Modern High Gain" delivers the typical fat sound that is rich in highs, similar to American amps like the Mesa/Boogie & Co. in 3 cascading amplification stages. Before the second and the third stages, the proportion of lows and highs is regulated, so that low tunings and quick passages are amplified without any "mud" and the signal always remains clear and assertive.
- **Lead:** In all preamp modes, the lead channel consists of an additionally cascaded tube circuit. In the "Classic" and "British" preamp setting, the lead channel may be viewed as more of a modification of "tuning" of vintage aspects. The "Modern High Gain" variant really gets going here. In this setting, VANDAL playfully takes on even drop-C or -D tunings.

Tip: Even if you don't want to use up all your gain in the crunch & lead channels, you can still try to go "a level higher" and take the (pre) gain level down a little for a slightly distorted sound. This increases the sound's complexity, and it will also seem more lively.

Pre & Post Gain

You can set up the desired amplification factor with the pre and post gain controllers. The pre-gain controller corresponds to the normal "gain" controller on most amps.

Using the post-gain controller, the currently active channel's overall volume may be adjusted to another channel, or a certain channel may be raised (e.g. for a solo).

Don't worry about switching settings around in the middle of things; the amplifier will remember the gain settings when channels are changed.

Voicing

There are a number of more or less open secrets with regard to the different signature sounds of guitar amps. This one might sound "bluesy", but doesn't sound so great with more gain, and the other is contoured for grindcore metal and fat sounds, but maybe it's a little anemic otherwise... How does this work?

Most amps in the real world use more or less the same or similar circuitry designs. The important thing is the number of amplification stages involved. Each new stage not only increases the level of audio complexity, it also has a large influence on how the way the signal is treated before and after that stage. Filtering is the key:

In addition to preset filtering circuits between the amplification stages, we gave VANDAL a little something we call "curve EQ". Assuming that you are using an EQ pedal before the amp, you can change the distorted sound drastically, because instead on the overall sound, you can favor certain frequencies. Curve EQ does something similar: it's located (in some cases in multiple instances) at strategic points between individual amplifier stages, and it filters the signal before it is distorted by the next stage. This is known as "voicing". For fun, turn a curve in both directions and move around a bit with the frequency controller in the spectrum. This will give the amp a completely different character. By combining channels, pre-amp modes and voicing, you can create a completely individual amp sound and even imitate other amps.

Equalization

The actual sound control (the "tone stack") functions rather conventionally: VANDAL offers low, mid & high settings.

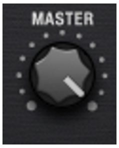
Everything functions like the passive sound regulation network in genuine amps so that the controllers influence each other to produce numerous variations. Depending on the preamp mode, the activation frequencies are slightly different.

Reverb

Genuine studio reverb devices are great, and VANDAL has two of them included in both multi-effect units. However, styles such as surf, twang, sixties, and others require a spring reverb. We've drawn on famous reverb spirals for our modelling.

Everything sounds natural with complete authenticity. This means that it has an old-school style rattle and typical rippling on the attack.

Master



The master control may be used to determine more than just the total amp volume. Since this is located before the end tubes in real amps, you can also use it to control the end stage distortion level.

This is very easily perceived in the class A mode; it doesn't just get louder, it also becomes more compressed and distorted.

If you are used to amps without a master volume, simply leave this at the maximum setting and control gain via a pre-stage and the final volume and the cabinet simulation or at the console output.

Bass Amplifier

During development of the VANDAL bass amp, it was our main goal to create a recipe consisting of the best amp designs in a single amp featuring an all-tube design that would be simple and easy to use.

The core features of the VANDAL bass amp are:

- Multi-stage tube preamp (with extra distortion stage)
- Contour control for extra-fat bass and glittering highs
- Controllable optical-electrical compressor
- 4-band sound control
- Class A/B tube end stage

Gain Control

The input level of the first amplification stage is set in the bass amp via the gain controller. Low to medium values leave the signal relatively neutral, and high values softly increase saturation.

Contour

The contour circuit is a filter stage that works similarly to the "Loudness" function. This control lets you give the bass sound an underlying character without additional EQ measures.

Turned all the way left, no effect will be present. However, the more the control is turned to the right, the more the mids will be thinned out and lows and highs raised. It's sort of like an "instant slap".

Comp (Opto Compressor)

After the contour circuit, the signal passes through the compressor stage (Comp). This is a simple but extremely musical "optical-electrical" design: the bass triggers a light source (e.g. an LED or a luminescent film) that is coupled with a photo resistor.

The louder you play, the more light falls on the photo resistor, which in turn dampens the signal. This may already be familiar to you from the most famous studio compressor for bassists, i.e. Urei LA2A, which functions according to the same principle.

There aren't that many settings to change in such a simple compressor. The control times are mostly determined by the sluggishness of the photo resistors. The comp control only sets the signal strength, which is routed to the light source.

Drive

After any possible compression, "Drive" provides the option to take the bass sound to the next level. The drive circuit is an additional amplification stage, which is proportionally mixed to the main signal. Signal distortion takes place depending on the frequency:

In spite of a level of distortion, the basses remain relatively clean and contoured. This guarantees complete "traction" of the signal when the controller is set all the way to the right.

Equalization

The following equalization stage offers 4 frequency ranges, in which case the two mid bands are variable.

Some bass amps allow their EQ circuits to have a drastic effect on the signal. We decided on a softer filtering with an edge slope of 6 dB/octave to avoid fully counterfeiting the essential character of the instrument in spite of the four bands.

Master

The final master volume controller sets the volume at the final stage. Similar to the guitar amp, the end tubes are also engaged in this case as much as remains sensible.

Through the interaction of gain, drive & master, countless saturations and distortion textures may be modeled as needed.

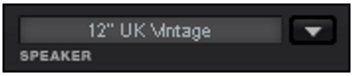
Cabinet Simulation

As was mentioned at the start, we went our own way in simulating the speakers and cabinets. In place of impulse responses from microphoned cabinets, we've calculated the individual components of this sort system:

- Loud speakers
- Housing
- Recording space
- Microphone

True to our "custom amplification" philosophy, this process provides total freedom and a certain measure of personal dynamics, which is the final deciding factor at the end concerning whether the system feels "lively". VANDAL also takes subtle effects into account, such as the reaction of the speaker's housing, the effect of this on the end stage that is applied, and so on.

Speaker selection



In the speaker selection list there are a wide variety of speakers in different sizes and with different sound characteristics. These are sorted according to guitar and bass types:

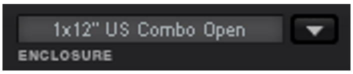
Guitar

- 10“ UK Vintage, 12“ UK Vintage, 12“ UK Modern, 12“ UK Blue, 10“ US Alnico, 12“ US Ceramic,15“ US Alnico, 15“ US Fat

Bass

- 10“ Custom Rock 10“ Custom Alu 15“ Custom Rock 15“ Custom Alu

Housing Selection



Various housing types may be combined with the available speakers in the housing list. This selection list is also organized according to guitar and bass types.

Guitar

- 1x12“ US Combo Open 1x12“ US Combo Closed 2x12“ UK 4x12“ UK 4x10“ US Tweed

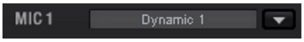
Bass

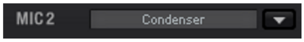
- 4x10“ Ported (Bassreflex) 1x15“ Ported (Bassreflex)

The housing has a significant effect on the resulting sound with a selected speaker. In general, open housing types are somewhat weaker on the lows, but go lower and sound somewhat more neutral than closed types. Larger housings also sound "bigger" and maybe "fatter". However, this is not always desired; for a guitar solo or for certain styles like blues, a small combo housing may be much more assertive or simply more "charming".

> It's up to your creativity whether you would like model the cabinets on realistic role models (e.g. 12" vintage speakers in a 4 x 12 housing) or place a 10-inch design in a 15-inch box. Refer to the "Advanced settings" a little further on for all other variations to help you create your own "personalised" speaker.

Microphones





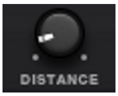
Cabinet simulation offers two separate microphones for the recording of the virtual speaker in a modeled recording space (MIC 1 & MIC 2). Similar to a real-life situation, you may position a virtual microphone stand in a space to achieve interesting mixing ratios and stereo effects.

For each of the two available microphones, you may select from a list which includes several popular studio types featuring characteristic sound models:

- Condenser, Dynamic 1, Dynamic 2

For each selected microphone, a range of sound design parameters is available:

Axis: The "on axis" (controller to the left) setting corresponds positioning of the microphone directly in the middle of the speaker cone. In this position, the sound will have many highs, but may sound piercing. For this reason, the microphone will often be positioned a bit "off axis" in order to achieve a softer sound.

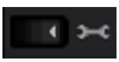


Distance: This control allows you to remove the microphone from the speaker and position it further away in the recording studio. This way you can make the sound more alive and loosen it up acoustically. Using two microphones with different panoramas (see below) enables you to create extremely realistic stereo images (note that the axis parameter has a decreasing effect in relation to the increasing distance).

Pan: This parameter distributes the microphone signal in the stereo sum at the output of the cabinet simulation.

Level: This adjusts the microphone volume.

Advanced Settings



Clicking the wrench symbol accesses additional cabinet simulation parameters:

Speaker Distortion: As discussed at the start of this manual, we have also modeled the non-linear aspects of a speaker. These dramatically change the sound, depending on how hard it is working. Under heavy loads, not only more distortion is created, but the dynamics and the frequency spectrum also change. This control allows you to set the effect of the non-linear behavior within certain limits. Please note that the acoustic effect of this parameter also depends on the amp's end stage.

Cabinet Balance: We have already explained that the housing is an important part of the overall sound. This parameter allows you to set just how much effect this has. Turned all the way to the left, the speaker exists almost "free in space", turned to the right, you are picking up sound directly "in the box".

Enclosure Damping: This parameter sets the level of housing resonance. The normal value is in the middle of the control range. When turned all the way left, the housing sounds "emptier" (increased resonance). Turned all the way right, the speaker sounds as if it were covered in insulating material, and typical resonance features are reduced.

Room Size: This adjusts the size of the recording space being modeled. The more the controller is turned to the right, the bigger the recording studio's dimensions become.

Room Damping: Similar to enclosure damping, the materials involved and the room's capacity to absorb may also be changed. When turned to the right, the room is dampened to the maximum.

MIC 1 / 2 Delay: Turning either knob (or both) delays the corresponding mic signal to up to 30 milliseconds. This can be handy in creating spatial effects (thereby taking advantage of the "Haas effect", allowing localisation through time delay). Interesting new tonal textures are also possible by using just very small amounts of delay on either mic.

Phase Flip: These switches regulate the phase length ("in-phase"/"out-of-phase") of both microphones. This is helpful in case the signal from VANDAL needs to be mixed with others or in order to create interesting sounds by changing the positions of the virtual microphones.

Rack effects (FX1/FX2)

There are effects that don't always work well when placed before the amp, e.g. reverb or delay, especially when they are distorted. Normally, these effects are better placed at the end of the signal chain.

For final processing and enrichment or whatever else you need, we offer two separate studio-quality effects units that behave just like real 19" rack devices.

Many of these algorithms create a stereo signal. As required, make sure that the channel strip is being calculated as stereo in the audio sequencer track.

Effects units may be selectively operated one behind the other (serial) or parallel. Switching may be changed via the mode switch.

The following algorithms & effects are available:

- **Mono Delay (msec & tempo sync):** Selectable as a simple delay with freely adjustable delay time or synched to the sequencer tempo with a musical raster. In case of higher feedback values, a reduction of the damping frequency is required to provide natural echo sound.
- **Stereo Delay (msec & tempo sync):** Features two models, just like mono delay. Repetitions may take place on separate channels (feedback controller to the right: dual delay) or in ping-pong mode (controller to the left), in which case the signal alternates between the sides.
- **Chorus:** Produces a typical "floating/shimmering sound" via modulated detuning of a signal to "thicken" the sound or spread it across the stereo field. Detuning is achieved via a short delay, the length of which may be varied via the modulation. This produces the so-called "Doppler" effect and broadens the signal.
- **Flanger:** Similar in terms of the algorithm that is applied to chorus, but different in that the delay time is significantly lower and delay works via repetitions (feedback). The flanger sounds more cutting and up-front than chorus.
- **Phaser:** A modulation effect just like chorus & flanger, but in this case no detuning takes place. Filter components periodically alter the signal's "phase response" (principle of the "phase shifter"). When mixed into the original, characteristic notches are produced in the frequency spectrum response (comb filter effects).
- **Room Reverb/Hall Reverb:** Reverb offers realistic simulation of realistic reverberation. Room creates the impression of a small to mid-sized recording room, while reverb produces the impression of a concert hall. A particular is that both effects algorithms provide a modulation parameter, which may remove

possible resonance at low dosages and can produce a soft chorus effect at higher values.
- **Vintage Plate Reverb:** The algorithms used in this type of reverb are quite similar to the ones that popular hardware effects units back in the 80s used to emulate that certain dense space of reverb plates. The resulting sound is correspondingly "wide", spacial and responds very directly, with immediate dispersion and no single echoes. Modulation of the reverb tail is also possible here, to minimize the typical ringing of the underlying metal-plate model.
- **LoFi:** Depending on its setting, this algorithm adds grit to the sound or a certain measure of signal destruction. Turn down the internal sample rate as much as you like to steal a few bits from the sound's resolution. Definitely an unconventional method...
- **Vintage Compressor:** Ideal for thickening up the signal a little. The algorithm emulates an older popular circuit design that is similar to studio legends like the Urei 1176 or simple compressor pedals. A so-called "FET building block" simply, effectively, and quite musically controls the volume via the input level, the set compression level, the compression ratio, and the response responses (attack and release).
- **3-Band EQ:** This sound controller works like a conventional mixer with a controller for the bass, the highs, and two controllers for the variable mids. This adds the final polish to your sound.

Console

In the quick start (view page 920) chapter, we mentioned that the console is the central switchboard for all of VANDAL's software. This interface is divided into various sections:

- Controllers for input and output levels, noise gate & limiter
- Management of all underlying components as a preset file
- Management of 5 remote controls for steering any parameters
- Chromatic tuner
- Advanced settings (MIDI, global)

Preset Menu

Presets are files within the VANDAL folder that consist of four scenes (but not all four must me used).

The default presets are organized in (sub)folders on the hard drive and may be located via the navigation structure in the presets menu.

An altered, but not yet saved preset is marked with an asterisk (*) in front of its name.

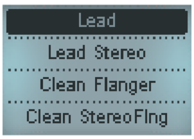
To save a preset, click on the diskette symbol and enter a descriptive name in the dialog that opens. Navigate to a subfolder or create a new one as required.

The number of possible presets available within VANDAL is unlimited. The same applies to variations to settings saved as new files. For the sake of preserving a clear overview, we recommend saving your own preset files in specific subfolders. The depth of the folder tree is unlimited.

Remote Parameters

The five controllers provide an elegant option for adjusting any desired parameter in currently activated components. These may be controlled via MIDI and via VST automation in the sequencer program. The remote control is also the interface between VANDAL and the outside world. The controls may be assigned via the advanced settings (view page 947).

Scene memory



Imagine you have to put together a set list for a gig. A good idea might be to create a list in Vandal in the form of presets.

It might be an idea to create a new folder and enter the location and date of the gig as a title. Then save the individual presets for each song in it.

As a guitarist, you will probably want more than a single setting for each of your songs. If more gain or volume is needed in a particular solo section, it wouldn't be very productive to create an entire preset for this.

The solution Vandal offers is a range of four variations for each preset, these are so-called "scenes". These may be switched using keys 1 - 4 beside the preset list or by clicking the names. All presets which are made from now on will only affect to the currently active scene. Scenes can be changed around at any time without discarding settings.

Settings may be transferred easily to other scenes simply by pressing the copy button for the source scene, and then switching to the target scene and pressing paste.

Copy & paste also works for presets, i. e. the clip board is not cleared when you change the presets, this enables settings to be saved easily into a new preset.

Scenes may be switched externally via MIDI (view page 949).

Chromatic Tuner

The chromatic tuner in the console has an automatic circuit which analyzes the recently played sound and shows the note and deviation from the idea (in cents) on the display. Simultaneously, the point moves up (too high) or down (too low) along the scale or stops in the center when the tuning is exact.

The arrows indicate the direction of the tuning (up or down).

The mute button switches the output signal from VANDAL to mute for as long as the chromatic tuner is activated (tuning fork symbol).

Advanced Settings

Clicking on the wrench symbol below the tuner opens a separate display.

On its right hand side, there is a sub menu for acessing in-depth adjustments of the software:

- remote ctrl (Control): Remote parameters for external control, either from the sequencer or via MIDI, can be assigned here.
- patch list: The underlying sub page serves for mapping MIDI or VST programs to actual presets. Because the preset library is a dynamic collection of arbitrary files on the hard drive, but the MIDI program scheme is tied to 128 possible programs, a proper mapping regime is needed. A "patch list" is used for that exact matter.
- preferences: Here, global settings can be made, affecting all instances of Vandal, regardless of the current preset.
- about: Shows the current installed plugin version and provides clickable links for the product web site and additional downloads.

Remote Control

You can use a standard MIDI controller in conjunction with Vandal to externally control any of the program's parameters. For example, preset changes present no problem either using program change commands, or using the wah-wah pedal via MIDI control change. A MIDI configuration may be set for each preset and each scene saved in Vandal.

A maximum of ten different control commands may be assigned at the same time (assignable in two rows of five). The source for controlling may be external MIDI, but the five remote controllers below the preset display are also available. An individual remote-controller/MIDI controller can control up to four parameters.

For each target parameter, a selection may be entered for the type and level of the control. For example, "Add to param" indicates that the value of the external controller or remote controller should be added to the parameter that is being controlled.

Let's assume that the controller, which is sending MIDI CC 7 (Volume), should also control the guitar amp's gain in the lead channel from zero up to the current control setting. Simultaneously, the master volume should be reduced to compensate for the volume. Proceed as follows:

- In the Remote 1 view click on MIDI learn in the MIDI controller range or select CC 7 volume from the list By doing this you have specified that you want to control the remote memory with your pedal.
- In the first target list, select "Guitar amp -> PreGain lead" as the target.
- In the list next to this, select "Multiply" as the control method.
- Move the fader right to the 100% mark. This indicates to the control that the depressed pedal should match the current controller maximum. If for example the gain fader is at 12 o'clock (i.e vertically) then a half-pushed pedal would give you a gain of 25%.
- In the second list, select "Guitar amp -> Master volume".
- Select Subtract as the method this time.
- Try setting the fader to 10-15%. Of course, the level of compensation depends on the concrete sound; if any at all... This is, after all just an example.
- Enter a sensible name for "Remote 1" by double clicking it, e .g. "Pedal gain".

The MIDI/Remote configuration may be left again by clicking the wrench symbol. The main display will now feature "Pedal gain" in the first remote storage section and the assigned MIDI controller (volume).

Patch List

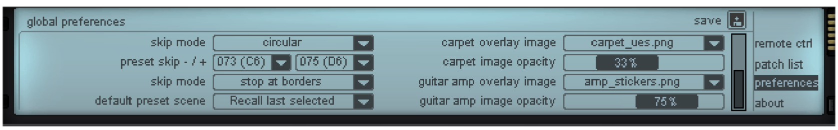
Because the preset library is a dynamic thing and can hold huge numbers of files (sheer unlimited), they can't simply be accessed by MIDI program change messages from outside. MIDI knows only about 128 programs, and it doesn't have a notion about the plugin's file organisation, for instance sub folders, or even multiples of those. For that reason, a "connection" between these two worlds is needed. That's where the patch list comes in.

This list is shown as pages of 16 programs / program slots. Each represents a MIDI/VST program.
Using the "edit" menu, the current page to edit is brought up. Alternatively, an external MIDI or VST program can be sent; the underlying page is automatically picked and the selected program is highlighted in the display. Presets can be simply assigned to programs by choosing an existing file from the corresponding menu.

A patch list can be saved using the disk icon. Just like the presets regime, multiple patch lists can be created and you can set up a library of different lists, each serving a particular purpose.
This can be handy for musicians gigging a lot, and changing their set very often. Using seperate patch lists makes programming an external controller much easier, as it only needs to be fed with the programs you really need for a show.

A selected / saved list can be stored as your preference, by hitting the "as default" icon. This is a global setting, affecting all Vandal instances. You'll find this setting again on the "preferences" page, labelled "startup patch list".

Preferences



On this page, numerous in-depth settings to the software can be accessed. These act as "global" modifiers, which means they affect all (following) instances of Vandal, and they work regardless of the current chosen preset.

- startup preset: Here, the default preset is specified, which will be loaded at plugin instanciation. Keep in mind that your sequencer program can freely override this preference, by sending a MIDI/VST program change command as last in the boot-up sequence, which might trigger a preset from a "patch list". So, this setting should probably be considered a fallback, in case no default patch is found.
- startup patch list: As mentioned earlier, this is the default list of preset files that get assigned to MIDI/VST program numbers.
- direct switch scene ← midi: The 4 scenes of a presets can be switched independently via MIDI. You can specify here, which scene is accessed by which command. Possible data types are either MIDI note number, or program change messages.
- scene skip -/+: for external controllers having up/down switching capability, proper control commands for skipping scenes can be chosen here.
- skip mode: You can specify, whether skipping should stop at either end, or whether further commands should result in a round-robin fashion (e.g. from scene 4 to scene 1).
- preset skip & skip mode: Just like with scenes.
- default preset scene: This menu specifies whether the first scene is activated upon preset load, or whether the preset should recall the one that was last selected by the time it got saved.
- input & gate level: Typically, the knob settings for the console's gate and input level are read directly from the current preset. Choosing "use current for all", you can specify global values as the reference. The values stored are taken directly from the current knob readouts.
- window resize: Normally, the two little buttons on the console's left side (below the "V" symbol) switch the stomp and the amp/cabinet/fx section on or off. This manual mode can be changed to automatic. Then, presets containing stomps and an amp will automatically maximize the window, while the view will collapse as soon as either the stomp or amp component slot isn't occupied. The window can also be forced to stay always maximised (so that the above mentioned console buttons have no function). The maximised mode might be helpful with some host programs that show dynamic window resize issues.
- mouse mode: Choose your favourite method here for handling knob movements. Knobs can either be turned in a linear fashion (up/down), or as a circular movement.
- tuner reference frequency: In case your music asks for a tuning other than standard 440Hz, the tuner can be recalibrated here.
- carpet / guitar amp overlay image: You can "skin" Vandal's stomp background (the "carpet" as well as the guitar amp's front cover (cloth) to your personal liking, by specifying own images.
  A "custom image" has to be in PNG format and reside inside the "_custom" folder (found inside the Vandal installation branch).
- image opacity: "Custom" images with transparency/alpha information are supported. Likewise, an image can be softly blended in or replace the background completely.

Cleaning/Restoration Suite

Only in Samplitude Pro X4 Suite or after optional activation.

The Cleaning & Restoration Suite consists of DeClicker/DeCrackler, DeClipper, DeHisser (view page 955), DeNoiser, Brilliance Enhancer and Spectral Cleaning (offline) (view page 966).

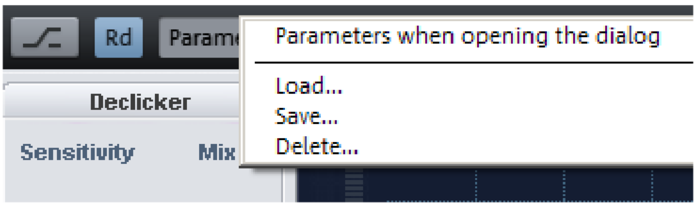
DeClicker/DeCrackler

The DeClicker removes crackling and single clicking noises that are typical of scratched records.

The DeCrackler algorithm was developed specifically to remove crackling noises, making it easier to remove crackling from old records.

The DeClicker is on the left side of the dialog, and the DeCrackler is on the right.

DeClicker/DeCrackler - presets

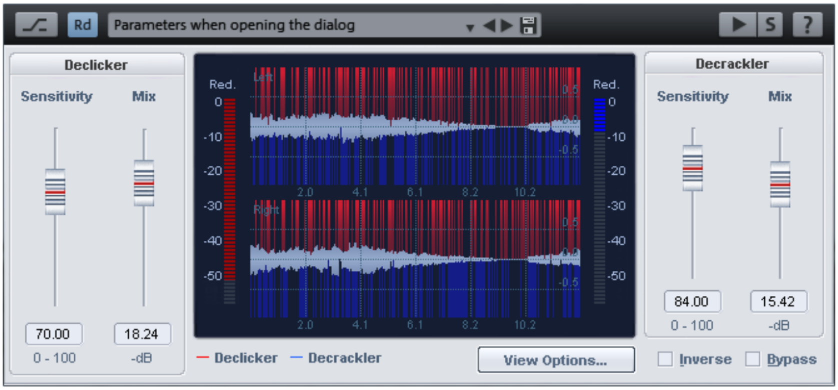


By selecting"**Parameters when dialog is opened**", you will undo all changes made in the dialog since it was opened. By closing the dialog, you will apply all current settings.

Save, **load** and **delete** functions are integrated into the presets list, where they are available in the lower section. The default file extension is ***.dck**.

DeClicker/DeCrackler – signal display

The signal display shows you the edited material as a **continuous waveform** or a **spectrogram**. In the display you can see at which positions the DeClicker (red) or the DeCrackler (blue) are active.



The left border of the graphic display features the reduction display for the DeClicker. It shows by how many decibel the detected click has been dampened. The right border features the reduction display for the DeCrackler. This indicates how many decibels the crackling has been reduced.

Display options: This opens a settings dialog to configure the signal display. Select the right and left channel or the middle from both left and right (L+R). You can additionally set the channel that should be visualized by both reduction displays. The settings are applied by closing the dialog.

DeClicker/DeCrackler – parameters and control elements

DeClicker sensitivity: This parameter determines the DeClicker's sensitivity to noise.

DeClicker dampening: This parameter influences the intensity at which the DeClicker engages at the detected areas in the audio material.

DeCrackler sensitivity: Use this parameter to determine the DeCrackler's sensitivity to noise.

DeCrackler dampening: Use this parameter to influence the intensity at which the DeCrackler engages at the detected areas in the audio material.

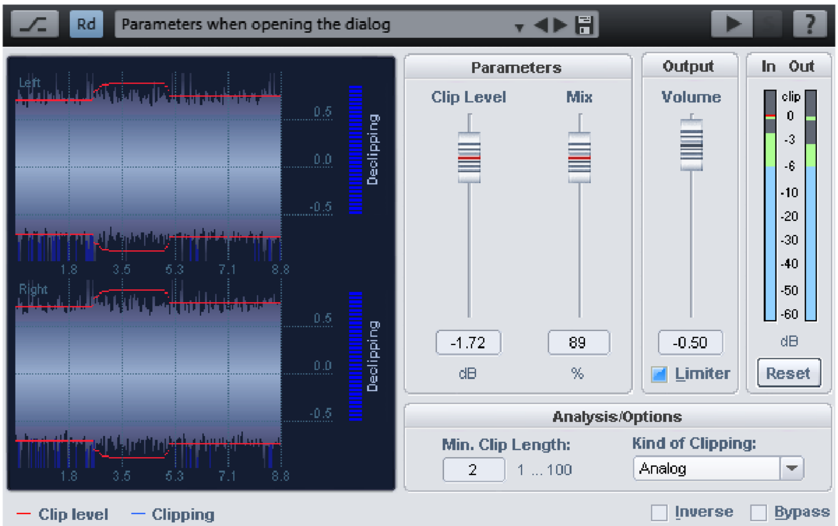
Inverse: If this switch is activated, only the part of the signal that is removed by the algorithm will be audible. If the parameters are set optimally, the complete distortion and just moderate components of the desired signal will be audible. If the correction parameters are too high, parts of the music or speech signal will also be filtered, which will discolor the music.

Bypass: The algorithm is removed from the signal route. This enables the unedited signal to be compared to the result of the algorithm.

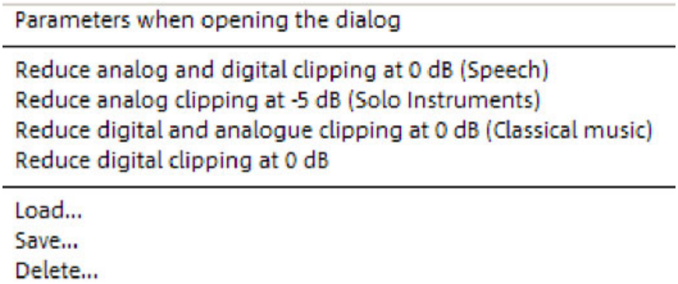
DeClipper

The DeClipper is a tool for removing overmodulation and distortions. Overmodulated passages are recalculated, or interpolated, based on the material immediately surrounding them.

The declipping algorithm is suitable for material featuring clearly audible overmodulation, e.g. distorted piano or vocals.



DeClipper - presets



By selecting"**Parameters when dialog is opened**", you will undo all changes made in the dialog since it was opened. By closing the dialog, you will apply all current settings.

Save, **load** and **delete** functions are integrated into the presets list, where they are available in the lower section. The default file extension is ***.dcp**.

DeClipper – signal display

The signal display shows you the edited material as a continuous waveform. The **Clip level** is shown as a red limiting line. Use the blue markers to see the positions where the de-clipper was used. The meter to the right of the waveform display also shows the intervention of the effect.

DeClipper – parameters and controls

Clip level: Enter the level at which the sample will be considered overmodulated and corrected.

Reduction: Use this parameter to set the intensity of the reduction in % at which the DeClipper alters the audio material.

Volume: This setting makes it possible to lower the output level, because the corrected output will tend to be a bit louder than the input signal as a result of waveform interpolation.

Limiter: This option limits the peaks that exceed the maximum level. Peaks higher than the max may result from interpolation.

Reset: Set the level to its original position here.

Analysis/options

Min. clip length: Set the minimum count of consecutive overmodulated samples for clipping detection.

Type of clipping: Optimizes clipping detection by setting the applicable setting for the material – "Analog", " Digital", "Analog + Digital".

Inverse: If is activated, only the part of the signal that is changed by the algorithm will be audible.

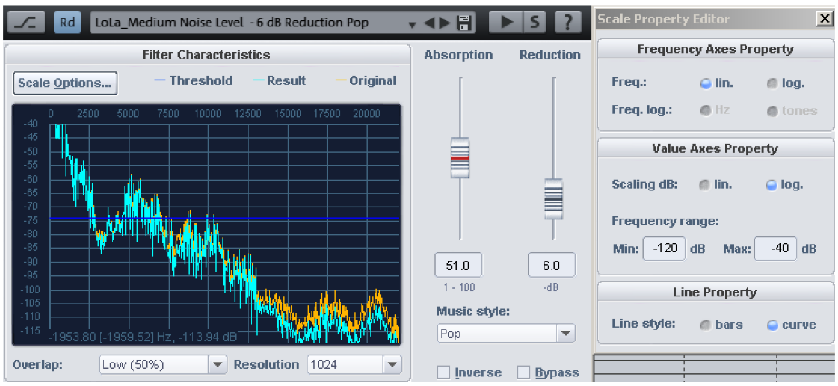
Bypass: The algorithm is removed from the signal route. This allows the unedited signal to be compared to the result of the algorithm.

DeClipper – sample applications

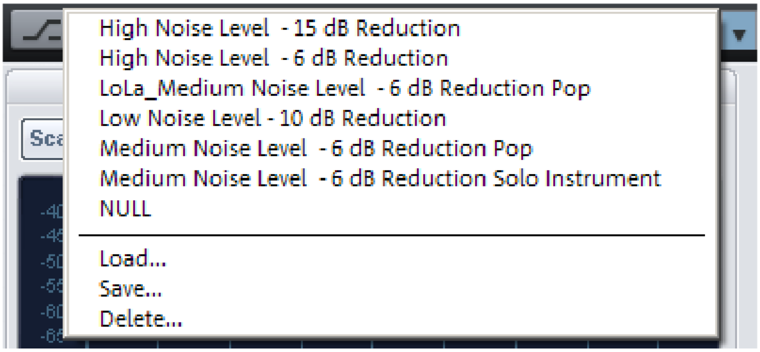
- Some DAT recorders have a protective analog switch, so that the level never reaches the digital mains (0 dB). In this case, a value of -0.5 dB or less is suitable.
- With an input of -6 dB (for example), all samples above half the control level are registered as overmodulated and recalculated. This allows a signal damaged by analog to be repaired.

DeHisser

This command opens the DeHisser for noise removal. The DeHisser eliminates regular, low-level "white" noise that is typically produced by recordings, microphones, pre-amplifiers, or transformers. Unlike during noise reduction, a sample of the distortion is not required.



DeHisser - settings and controls



The **load, save** and **delete** functions are integrated into the preset list, where they are always available. The default file extension is *.deh.

Correction (1-100): With this parameter a threshold value will be set for differentiating the signal from noise. The right setting for this parameter is crucial for good results. In comparison to noise reduction, where this setting will be automatically determined from the noise sample, with the de-hisser, you have to determine the noise level manually in relation to an absolute noise level.

Low settings can lead to incomplete noise removal or increased artifacts. High settings can cause discoloration of the original sample, phase-like distortion or chirping modulation effects. The louder the noise, the more precisely your should try to set the DeHisser.

Noise reduction (0-30) dB: Set the noise damping here. The highest possible damping value is 30 dB.

In practice, extremely quiet passages, like the slow fading out of one instrument prove to be critical – here the noise level can exceed that of the audio signal.

Limit yourself to one low noise damping (-10 -15 dB) to avoid side effects like artifacts.

Music style: Minimize artifacts by selecting a fitting music style to aid in determining transients in the track and by de-hissing only between them.

Inverse: If this switch is activated, you will hear only the part of the signal that is removed by the algorithm. If the correction parameters are set optimally, only the noise will be audible. If the correction parameters are too high, parts of the music or speech signal will also be filtered, which will discolor the music.

Hint: If "inverse" is selected, the damping parameter will have no influence.

Bypass: The algorithm is removed from the signal route. In this way, the unedited signal can be compared to the result of the algorithm.

Overlapping: High values for overlapping allow the de-hisser to follow the signal quicker. When low values are selected, the algorithm is less quick when adjusting to the signal properties. The settings "Medium" and "High" are useful for most standard cases. "High" is most effective when applying the De-hisser to mixed (multiple instruments) tracks.

Rate: This parameter sets the rate (low, medium, or high) used by the noise reduction algorithm. Experiment with this parameter; a higher rate does not always guarantee the best results.

As an example, Low resolution often works best when applying noise reduction to short percussive sounds and speech. Instrumental music with slower tempos high resolution.

Scaling options

Frequency display Freq lin/log: Linear display of frequency provides a more detailed view of the high end. Logarithmic display of frequency provides a more detailed view of the low end. The logarithmic display corresponds to the human perception of volume. Like on a keyboard of a key instrument, here the graphic spacing of the entire area is uniform throughout. If you have selected the logarithmic view of the frequencies, you can select between frequency and note representation for the horizontal axis.

Value display dB lin/log: Here you can choose between a logarithmic and linear level scaling. The logarithmic display corresponds to the human perception of volume.

Value range min/max dB: Enter the level range for displaying the real-time spectrum.

Curve depiction: Choose to display the original and the corrected frequency progression as a curve or as a bar diagram.

DeHisser – graphic display description

The vertical axis to the left indicates the level of the spectrum; choose between frequency or notes for the horizontal axis.

The yellow curve displays the original spectrum of the signal. Simultaneously, the first second of the selected audio segment of the active segment will be spectrally analyzed.

The light blue curve represents the spectrum after manipulation by the algorithm.

The dark blue curve represents the threshold of the "correction" parameters.

If the threshold is higher than the level of the spectrum (i.e. the dark blue line is above the yellow curve), the signal for De-hissing will be filtered out at these frequencies.

DeHisser - artifacts

The DeHisser was created especially for use with mastering in order to remove uniform and low-level hissing, which allows music and speech signals remain as pure as possible.

The algorithm is only partially suited to removing sound distortions featuring levels that reach or exceed the original signal. A metallic chirping or twittering sound may appear, the so-called artifact noise. In these cases, use the DeNoiser (view page 959) algorithm to remove noise and distortions.

DeHisser - the best setting

1. Search your audio material for a critical segment to preview. Critical segments are the softest parts of the music or speech, where the noise has a comparable level to that of the signal.

2. Set the damping parameter to the highest value (-30 dB).

3. Slowly increase the value of the correction setting.

There are four stages in this process:

Stage 1: If the value is very low, no noise will be removed.

Stage 2: The noise is partially removed. Depending on the level of the noise signal, a small number of artifacts may be introduced.

Stage 3: The noise is completely removed.

Stage 4: If the value is very high, not only will the noise be removed, but also a part of the signal. Press "Inverse" to hear this. The audio material will lose its brilliance and sound duller.

Optimal settings are usually available in **Stage 3**, where the noise signal will already be removed, but the audio signal is played back unaltered.

Tip: When monitoring the changes, it often helps to use a high monitoring level on the mixing board amplifier. It is recommended to use a system with a high signal-to-noise ratio. Monitoring using headphones is also helpful.

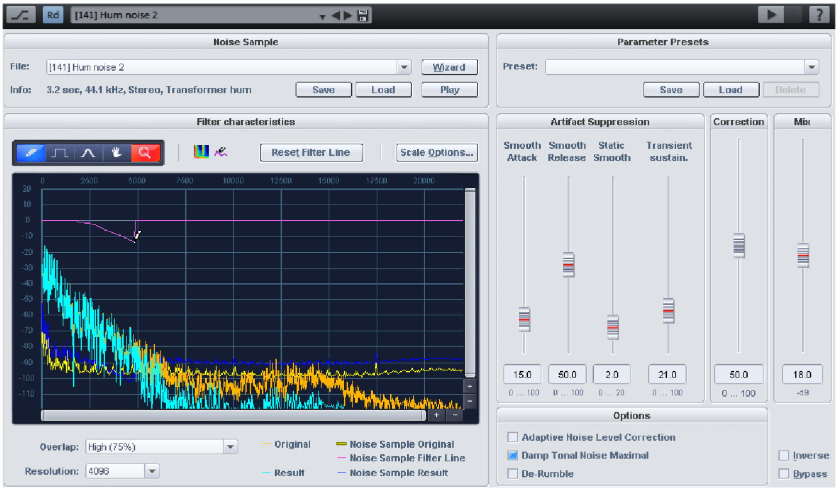
4. If the noise signal cannot be removed without discoloring the music or speech, reduce the damping parameter value until a compromise is reached between noise reduction and an acceptable discoloration of the audio material.

Tip: In the case of very high noise levels, try first to remove the noise from single tracks. Dehissing the complete mix requires more careful manipulation.

To remove humming noise, please use noise reduction (DeNoiser) (view page 959).

DeNoiser – remove unwanted noise

Use the noise removal function to free your wave projects/objects of unwanted noise without any noticeable reduction of the sample's range. The algorithm requires a noise sample beforehand to do this. The function works especially well to remove consistent, long-lasting noise like ventilation noise, artifacts from low-quality sound cards, tape deck noise, or feedback. Although the algorithm has not been specifically developed to remove clicking crackling, it is possible to get good results using it to remove this type of noise from old records.



DeNoiser – quick start

1. Select an area in the selected object or in a wave project of a virtual project that only features the noise.
2. Copy this area using the "Effects -> Restoration -> Get noise sample" into the "Noise sample" project. Especially in the case of hissing, better results are usually achieved by the algorithm the longer the noise samples are. If the length of the noise sample is longer than one minute, then there will only be a marginal improvement to the sound.

3. Select the range of the original sample in the wave project where de-hissing should be applied or select the corresponding object in the VIP.
4. Open the De-noiser via "Effects -> Restoration -> DeNoiser".
5. If the sample selection list for the noise sample (under "Noise Sample -> File") is not already set to "Noise Print", rearrange the list entry accordingly.
6. Press the "Play/stop" button for the real-time preview function. Once you are satisfied with the result, press "OK". Otherwise, change the parameters.

DeNoiser - Noise Sample

File: Select a group of noise tests in the selection list. The noise samples displayed in the list are located in the "Noise sample" directory or are temporarily available noise prints. If the effect reference file is an automatically created copy of the original noise sample, the reference file name appears at the end of the original name.

Info: Provides information about length and type of the set noise sample.

Save: This button saves the currently set noise sample as a file for later use.

Load: This buttons opens a file selection dialog to import a wave file as a noise file.

Assistant: Opens the **"Noise print assistant"** dialog to extract a noise sample. Learn more more about this assistant below (view page 965).

Play: This command plays back the recently set noise sample.

DeNoiser – parameter presets

Load, **save**, **delete** presets: Save, load, or delete settings here. The noise sample is not included. The default file extension is ***.nrp**.

DeNoiser - filter settings

Pencil function: Drawing changes in the graphical display using the mouse tool edits the purple curve. This curve filters the noise sample.

There are 5 tools to help draw and navigate in the signal display:

- Freehand drawing with pen tools: Hold down the "Shift" key to draw straight lines.
- Drawing pencil for quantized drawing in predetermined decibel steps. The dB steps depend on magnification of the filter curve in the value display. Hold down "Shift" to draw straight lines.
- Reshape tool for bending curves: Through repeated clicking, the curve bends away from the mouse tool. The further the mouse tool is from the filter line, the wider the bend will be. If "Ctrl" is also held down, this effect will be even stronger.

- Navigation tool: Use this tool to navigate the coordinates system in the range while at a zoom stage. If you have not zoomed into the display, the navigation tool, the graphical display will not be affected.
- Zoom tool: Click with the left mouse button to zoom into the display; zoom out with the right mouse button.

In addition to the drawing tools, two additional buttons are available to you.

Activate spectral view: By clicking this button, **the signal will appear as a spectrum**. The areas of the spectrogram influenced by the De-noiser will appear red.

Activate direct mode for valuation curve: The second tool activates **"Direct" mode for drawing freehand curves**. The settings will be recalculated into a relative filter curve in the background.

Reset filter curve: This function sets the filter curve to its default settings.

Scale options: This button opens the "Scale options" dialog to adjust the signal display and the filter curve's value range as required.

Signal display: The graphic displays the **original spectrum** of the distortion **(yellow curve)** and the **corrected spectrum** which the algorithm uses internally to remove distortions **(blue curve)**. The level of the corrected spectrum is set via the **correction** fader. The curve can be smoothed via the **static smoothing** parameter. If this effect is active (for example during playback or monitoring), you can monitor the input and the output spectrum.

The **violet line** provides **filtering for the noise spectrum** (see above). Filtering effects the display of the corrected spectrum (blue line).

Overlapping: The exactness of the algorithm may be selected here in several steps. Processing times increase with increasing overlapping as the quality of the results improves.

Resolution: This parameter sets the rate (low, medium, or high) used by the noise reduction algorithm. The higher the resolution, the longer the algorithm will take to process. Experiment with this parameter, since a higher rate does not always guarantee the best results. For example, low resolution often works best for noise reduction applied to short percussive sounds and speech. Low-frequency sounds require adequate resolution in the low-frequency range; this may be achieved by setting this parameter to at least 4096.

DeNoiser - artifacts

When editing sound distortions with levels that reach or exceed that of the original signal, the algorithm may leave behind a metallic chirping or twittering sound, the so-called "artifact" noise. Its level is much lower than that of the original distortion, usually at around -20 dB, but due to its synthetic origins, the ear reacts sensitively to it.

DeNoiser – parameters

Artifact suppression

These settings suppress artifacts as they occur in broadband distortions of higher levels. Note that when higher values are selected, the quality of playback result may decrease.

Smooth attack: This parameter controls the attack during noise reduction. If delay is high, artifacts will be suppressed more effectively. The time ratio or impulse power of the audio material may decrease, however. In case of speech or singing, high values will not always produce the best results. This method of artifact suppression is recommended for orchestral instrument.

Smooth release: Use this parameter reduce an overly strong reduction of the release phase. Artifacts may reappear when higher values are used, however. Look for an acceptable compromise between artifact suppression and impulse response.

Static smoothing: This function smoothes the corrected noise spectrum used internally by the algorithm to remove noise. Smoothing of the blue curve may be visually confirmed by looking at the graphic. The artifacts are therefore reduced. In speech, vocals or pop music, higher values will usually provide good results. For orchestral instruments, high value settings of this parameter may cause unpleasant roughening of the sound

Consolidate transients: Use this parameter to influence the algorithm so that the distortion removal works more contained with transients. For example, in case of noisy jazz or pop recordings, this results in a clear improvement.

Correction

This parameter raises or lowers the noise spectrum (dark blue curve). Low settings may cause incomplete noise removal or increased artifacts. High settings can cause discoloration of the original sample, phase-like distortion, or to chirping modulation effects. The occurrence of these effects is dependent on the type of noise as well as the consistency of the original material. A higher noise level does not necessarily require a higher correction level.

Damping

Set noise damping in dB by entering a value between 0 dB and -40 dB.

In many cases, it is desirable not to remove noise completely: for example, when working with gramophone recordings, it can be desirable to leave some of that "gramophone feeling" in. Background noise from "on location" reports does not need to be removed completely. When noise is not completely eliminated, the occurrence of artifacts or discoloration is reduced. Think of the process in terms of reduction, not 100% elimination.

Options

Adaptive correction: Activate this switch to achieve time-adjustable, automatic customization of the value for the "Correction" parameter. While the DeNoiser is active, you can the monitor the constant adjustment of the "Correction" fader in the input field. "Adaptive correction" is useful for distortions with variable noise levels.

Maximize tonal distortion dampen: Removes tonal distortions such as humming or camera sounds; activate this button to remove this unwanted material. The "Dampen" parameter only affects the non-tonal portions of the signal (hissing). This may produce better results, since the dampening of tonal signal portions creates fewer artifacts than the dampening of non-tonal distortions. The resolution value should amount to at least 4096.

De-rumbler: Frequencies below 40 Hz will be considerably dampened. This way impact noise like footsteps or rumbling on records can be removed.

Inverse: If this is activated, only the part of the signal that is removed by the algorithm will be audible. If the parameters are set optimally, you will hear the complete distortion and just moderate components of the desired signal. If the parameters are not set correctly, larger components of the music or spoken signal will be filtered, which may cause discoloration of the sound. However, the part of the artifact resulting from the remaining components of the distortions is hardly decipherable by monitoring the inverse signal.

Bypass: The algorithm is removed from the signal route.

DeNoiser - the best setting

To find optimal settings for noise removal in the material, proceed as follows:

1. Search for a good "**Correction**" setting. The distortion sound should no longer be audible. If artifacts occur, they shouldn't be suppressed by setting a very high correction value. The result may sound lackluster.

2. To increase artifact suppression, increase the values for **smoothing** and/or **static smoothing**. Which parameter is most useful depends on the nature of the audio material from which the distortion is to be removed.
3. Try to slightly **reduce** the **correction**. Artifacts will increase in this case. Now raise the "**Smooth attack**" parameter and/or "**Static filter smooth**". Use the slider to improve your results.
4. A high degree of **overlapping** will increase quality.
5. Drag the **"Preserve transients"** parameter as far out as possible until noise increases and the distortion becomes audible at the transients (with hissing, this effect is recognizable by the type of hissing modulation). Reduce the level again slightly.
6. For difficult cases, we recommend the **freehand draw filter curve for the noise sample**. Problematic areas may be **influenced by increasing or decreasing the distortion spectrum**.
7. If there is no audible improvement, try correcting the noise instead of completely removing it. This may be attempted by reducing the value for mix **dampening**.
8. Whether or not artifacts are audible also depends on the monitor volume and the frequency response in the playback. If you know how your work will be reproduced (cinema theater, on TV, radio, etc.), you should work under corresponding monitoring conditions.

DeNoiser - Problems and Solutions

Problem: The result is silence

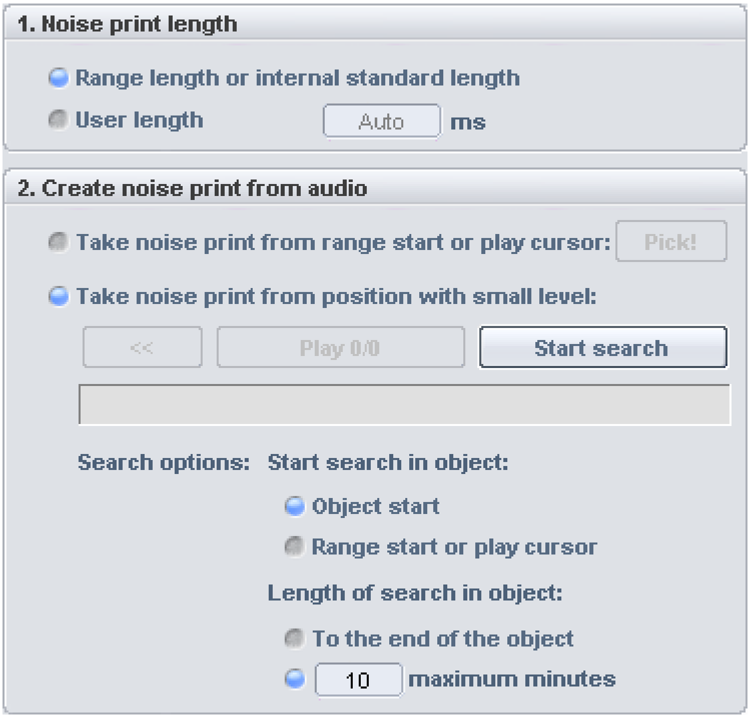
Solution: It is likely that the noise sample inadvertently contains the original signal. Use another noise sample.

Problem: Strong distortion, loud tweeting, strong damping of the used signal

Solution: It is likely that the sound sample contains sections from the original signal or from another sample. Use another noise sample.

DeNoiser - noise print assistant

The noise print assistant offers (in addition to the "Get noise sample" menu) two options to create a noise sample, whereby the "Pick" method of the "Get noise sample" function may be the better approach.



Proceed as follows to create a noise sample using the noise print assistant:

1. Determine noise print length

Range length or internal standard value: If a range has been stretched, a range length will be used. Ensure that the range is not longer than necessary, since every created noise sample instance in real time will be added to the project. If, conversely, no range is selected, an internal length is used, which will normally deliver useful results.

Target length: Set an explicit length for the noise sample.

2. Extract noise print from the audio material

Take noise print from range start or play cursor: The noise sample will be taken from the selected range, if available; otherwise, the standard or user defined length is used, starting from the play cursor position.

Take noise print from position with small level: This search function analyzes the audio object for soft areas. Set the search area above the search options and click the "**Start search**" button. Cancel the search by pressing "Esc".

After at least one quiet range has been located, select it using **<<** and **>>** and listen to it using "**Play...**"

Search options:

Start the search in object – object type: The search area starts at object type

Start the search in object – range start or the play cursor: Search area starts at the start of the range if you have defined a range beforehand. Otherwise, the search will begin at the play cursor.

Length of search in the object – until object end: The search for soft passages stretches until object end.

Search length in object – x minutes maximum: Enter the search length in minutes here.

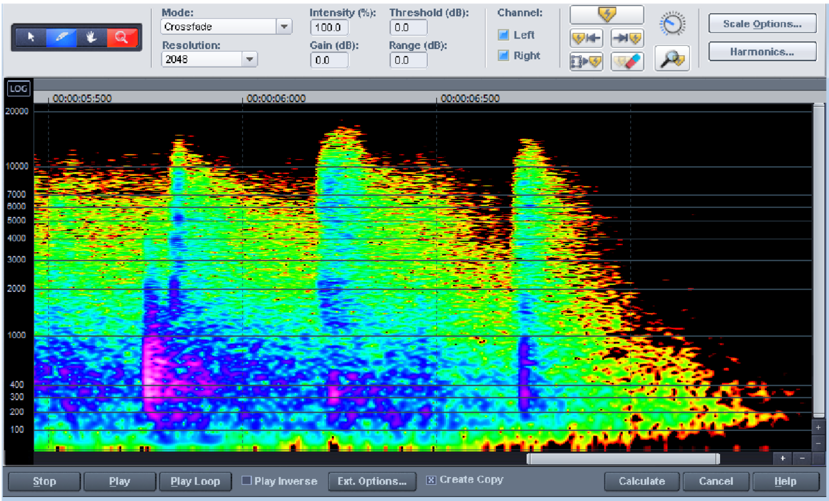
3. Exit the assistant with "OK"

Spectral Cleaning (Offline)

Spectral Cleaning – the basics

With Spectral Cleaning you can remove distortions (like coughing, whistling, or singular claps) from a recording without influencing the wanted signal.

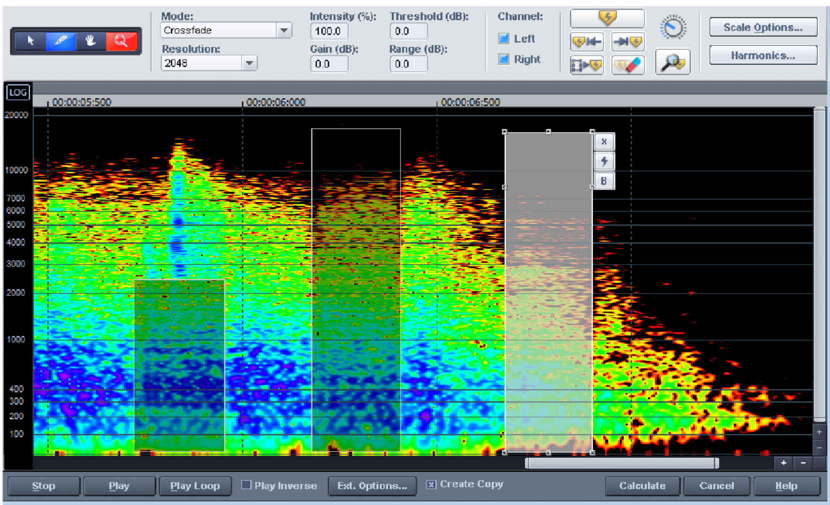
The graphical interpretation of the music can be seen in the Spectral Cleaning Editor via a spectrogram. This displays the frequency proportions in a time curve. The volume of frequencies is visualized via a color code or via its brightness.



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

A continuous sound is displayed by a pattern consisting of horizontal lines, which correspond to the sound components or overtones of the sound. A distortion with an impulse quality is seen as a vertical line.

In Spectral Cleaning you can highlight interferences with the drawing tool before removing them. So that there is no audible gap, removed components of the original frequency spectrum from the wanted signal that surrounds the distortion are recalculated into the recording.



The Spectral Cleaning is useful for removing all short, impulsive sounds in the music like popping, coughing, or even short dropouts. For permanent noises like humming, etc. use the DeHisser (view page 955) or the DeNoiser (view page 959).

Spectral Cleaning – operation

Select the object and set the play cursor to the rough position of the distortion. Next, open the Spectral Cleaning Editor via the "Offline effects" menu, or via the object context menu under "Effects (offline)".

Select the drawing tool. The mouse pointer will turn into a pen icon. Highlight the nose by drawing a rectangle around it using the mouse. You can also highlight and remove multiple interferences in the editing window.

If you move a noise selection with the selection/draw tool, then you can **lock out vertical movement on the frequency axis by holding down "Alt"**. Holding down the **"Shift" key also locks out horizontal movement** on the timeline.

"Ctrl + X", "Ctrl + C", "Ctrl + V" may be used to cut, copy, and insert selections, respectively. This only copies the settings of the rectangle, and not any audio material.

If you set the cursor in front of the selected distortions and press "Play", you will hear the result of the noise removal in the preview function. You can modify the frame around the distortion selection using handles to optimize the signal.

Three small buttons visible at the position of your selection.

Delete (x symbol): Remove the selection by clicking this button.

Show distortion (lightning symbol): After marking the noise signal with the drawing tool, the result of the noise removal will normally be displayed in the selection. The "Show original" button displays the original signal for comparison purposes.

Bypass (B): Once you have drawn in the selection, you can already monitor the results of noise removal during playback. If, however, you do press the "Bypass button, you will continue to hear the original signal.

Sometimes it's necessary to modify the effects parameters (see "Spectral Cleaning - Edit modes"). If you're happy with the result, press the "Calculate" button.

Spectral Cleaning – tool bar

You can assign various tools to the left and right mouse buttons, by clicking on the corresponding tool. The tool for the left mouse button is displayed in blue and the right mouse button in red.

Selection tool: Use this tool to modify the existing selection window.

The mouse pointer will turn into a double ended arrow when you hover over the edges of the selection rectangle, you can drag the edges and adjust the selection size.

If you are using the selection tool inside a frame the mouse will turn in to a crosshair. Now you can position the entire selection frame in the Spectral Cleaning window however you like.

Drawing tool: This tool allows you to change the selection size (just like with the selection tool) and position the selection window anywhere.

You may also drag out any number of selection windows with this tool to remove noise.

Note: Double clicking on the selection applies vertically it to the entire frequency range.

Navigation tool: This is useful when you are working with a high zoom factor and can't see the entire object contents in the Spectral Cleaning window.

With this tool you can move the window section displayed by holding the left mouse button down.

Zoom tool: After selecting this tool you can left click to zoom into the window.

By clicking and dragging with the left mouse button held down, you can stretch the zoom area in the Spectral Cleaning window. With a right-click you can zoom out of a selection.

Spectral Cleaning - Editing modes

With the "Mode" setting you can specify how you want to remove the distortion. All marked interference will be edited with the selected mode independent of the "Level" fader.

The following modes are available to you:

Crossfade: This mode replaces the noise with the signal from the areas directly bordering the selection.

Crossfade (hard): This mode behaves like "Crossfade" but has a more powerful effect. In this way you can eliminate particularly noisy interference more effectively. It is especially important to mark the interference precisely when doing this.

Crossfade (from left): In this mode the noise is replaced by the audio signal located directly to the left of the selection.

Crossfade (from right): In this mode the noise is replaced by the audio signal located directly to the right of the selection.

Gap: This mode is especially useful for very short dropouts.

Dampen: This mode dampens the selected noise. Surrounding audio material will not be included. In this way no artifacts will be created, but there may be unexpected silence or even a drop-out. Use this mode if a distortion is present in a very narrow frequency range or if the surrounding noises are very quiet.

Fade In: This mode is similar to the "Dampen" mode. However, here the signal is faded in instead of uniformly dampened. This means that frequency-selective fade-ins can be created.

Fade Out:: In this mode the selected noise will be faded out across the selected area on the timeline. In some cases you can use the „Hide“ option, e. g. for noise at the end of a track.

Level: Use this control to specify the degree of change. Moving the fader to the far right corresponds to a 100% level of change.

> **Note**: The "Edit selection" and "Level" settings can be controlled for each selected area individually, or as a multiple selection using "Ctrl".

Resolution: Determines the resolution of the applied FFT (Fast Fourier Transformation). Lower values often produce a more effective removal of the disturbance, but result in more tonal artifacts. They are also suitable for short interferences with powerful transients, such as clicks. Higher values are recommended for longer noises such as coughs etc. For optimal results first try editing an area with a low resolution, remove a transient part and then eliminate the rest of it with a higher resolution.

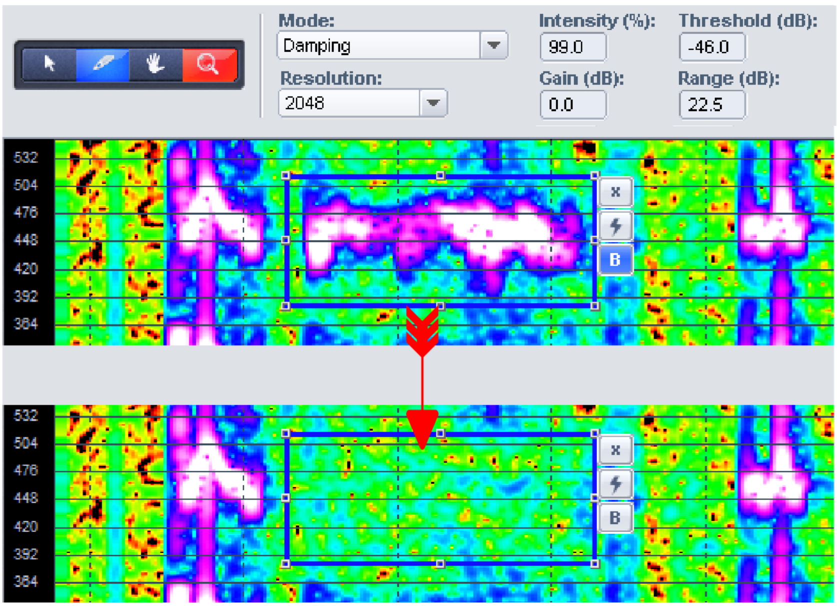
> **Note:** The setting "Resolution" applies to all edits and cannot be customized for each selection.

Gain (not in Samplitude Pro X2 Suite): This can be used to boost the level to compensate for a drop in volume caused by interpolation. This is particularly suitable for removing gaps when editing multitrack recordings.

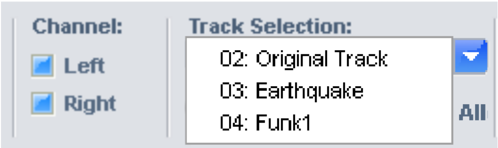
Threshold Value (not in Samplitude Pro X3 Suite): This setting can be used to limit the interpolation to a specific amplitude of a signal in the spectrum. This means certain notes can be removed from the spectrum without influencing background noises.

> **Note:** This function will only be effective with a set area and should be used in the editing mode "Dampen" where possible.

Range (not in Samplitude Pro X6 Suite): This range determines the volume window in which interpolation of the set threshold can be performed. Quieter or louder signal components outside the area will not be affected.



Channel left/right: These buttons show you the corresponding channel in the spectrogram. If the interference only affects one channel, you can switch off the other channels leaving them unaffected.



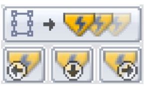
Track Selection (not in Samplitude Pro X6 Suite): If there are objects from different tracks selected, you can specify here for which track the calculation should be performed.

Notice: To select more than one object, the objects have to be of equal length and start at the same time.

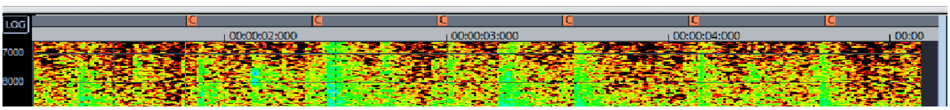
Apply to All Tracks (not in Samplitude Pro X6 Suite): With this option you define, if the calculation should be transferred to the selected single track, to all tracks of a previeously defined group of tracks or to all tracks.

Spectral Cleaning - Click-Marker

To highlight special points, quickly jump between or delete markers in the spectrogram, use the so called "Click marker" buttons in Spectral cleaning interface.

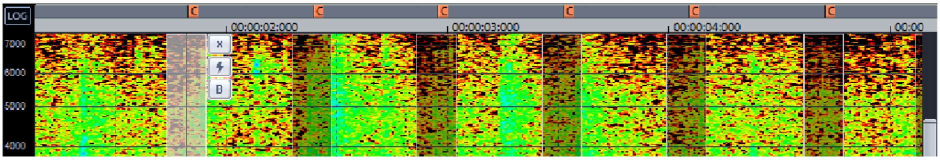


You can set and delete click markers. Created click markers are marked with a "C" in the editing window's marker bar.

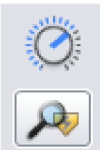


With "Jump to next marker" (Shortcut: +) or "jump to previous marker" (Shortcut: -) you can navigate quickly between the markers in the Spectrogram. To move among the click markers only and not the standard markers, hold down "Ctrl".

By pressing the "Apply range selection to all click markers" button, you copy the current selection frame to all positions where there is a click marker. This allows you to use the settings of the current markings on all click marker positions. This is particularly well suited for removing interferences.



Not in Samplitude Pro X6 Suite: You can use the button „Automatic Search for Click Markers“ to search through the entire project for clicks and mark them so they can be edited together. With the corresponding knob you can set the sensitivity of the search, i.e. the higher the level the more clicks will be found.



Note: Select suspicious areas in the arranger before opening the Spectral Cleaning dialog. To transform a standard/beat/tempo marker into a click marker in the spectrogram, click the corresponding marker in the Spectral Cleaning marker bar. Next click on the button "Set Current Marker". Now you can click again on the marker position to display the created click marker.

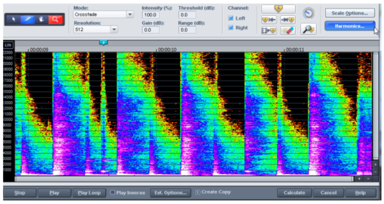
Harmonics

(Not in Samplitude Pro X6 Suite)

You can remove individual noises, e. g. 50 Hz buzzing, as well as harmonic overtones from a spectrum using this function. To do this an individual sound is selected in the editing window with the smallest possible edge. Once the calculation is activated for harmonics, the corresponding harmonics will be automatically provided with an editing rectangle in the editing window above the selected root.

Changing the root always leads to a change in the harmonics. In the settings dialog you can determine whether or not harmonics should be observed during calculation and if so, which ones in particular.

Using the option "**Continue harmonics**" it is possible to subtract out those harmonics that are above 32.



Edit harmonics: If active, the settings of all dialogs for calculating harmonics are included in the calculation.

Level (%): Customized selection of which harmonics are to be observed during calculation. The "Level" which is used can be determined for each individual harmonic. You can adjust the settings for the root (1 harmonic) to up to 32 harmonics. In addition you can use the "Harmonic continue" option to include harmonics in the calculation.

Even/Odd: Selection of even or odd numbered harmonics.

Fundamental frequency: This displays the fundamental frequency of the bottom most selection in the editing window and can be changed by entering a value or by moving it in the editing window.

Absolute/relative bandwidth: In the "Absolute bandwidth" the height and therefore frequency range of the harmonics selection rectangle is the same as the selected root. The frequency range increases in size with each harmonic in relation to the harmonic's frequency if the bandwidth is set to "Relative". This is so that relevant acoustic ranges can be achieved properly.

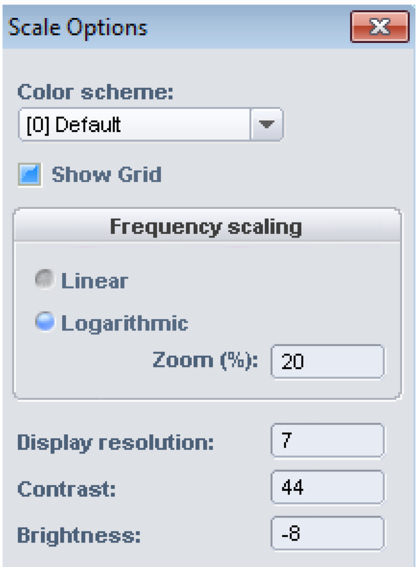
Continue harmonics: With this option any harmonics above 32 harmonics are included in the calculation with a linear dropping curve in order to achieve a thorough removal of all possible harmonics.

Uniform level: This means that the level set for 1 harmonic is also used for all other harmonics.

Edit between harmonics: If this option is activated, Spectral Cleaning will edit the sections between the selected harmonics. If you deactivate one or several harmonics, your sections will be edited up to the next active harmonics.

Scale Options

In this dialog you can adjust the settings that deal with color and content display.



Color scheme: Here you can choose a predefined color scheme which is used to display the spectrum,

Show grid: Displays a time and frequency grid in the spectrogram.

Frequency Scaling (not in Samplitude Pro X6 Suite): Here you can switch the display of the spectrum from linear to logarithmic. This is recommend for editing low frequency ranges. So that low frequencies are triggered higher up yet still display the entire frequency spectrum. The logarithmic ordering can be used to adjust the logarithmic display to meet specific requirements more precisely.

Note: Frequency scaling can be switched between linear and logarithmic using a button in the editing window.

Display Resolution (not in Samplitude Pro X6 Suite): This setting allows you to display the spectrogram in another resolution in order to achieve either a better spectral or chronological resolution for the range of interference.

Note: Calculation happens independently of the selected display resolution and is performed with the set resolution in the spectral cleaning dialog.

Contrast, Brightness (not in Samplitude Pro X6 Suite): Adjusts the contrast and brightness of the spectral display, e. g. to make very quite audio material visible.

Spectral Cleaning - Playback section

Stop: Stops playback (Keyboard shortcut: Space Bar)

Play: Plays from the playback marker position (Keyboard shortcut: Space Bar)

Solo: Plays the edited track solo (Keyboard shortcut: S)

Play loop: Repeats playback within a selected range (Keyboard shortcut: L)

Play Inverse: Plays back the differential signal, i.e. the removed sounds (Keyboard shortcut: I)

Mixer (not in Samplitude Pro X6 Suite): Opens the mixer (Keyboard shortcut: M)

Advanced options: Opens the settings dialog for destructive effect calculation

Create copy: Creates a copy of the current file during calculation

Calculate: Executes calculations and closes the dialog (Keyboard shortcut: Enter)

Undo (not in Samplitude Pro X6 Suite): Undo slice edits.

Redo (not in Samplitude Pro X6 Suite): Redo slice edits.

Leave Dialog Open (not in Samplitude Pro X6 Suite): The dialog remains open after the calculation is complete

Close: Closes the dialog window

Help: Opens the "Help" file.

Spectral Cleaning - Shortcuts

(Not in Samplitude Pro X6 Suite)

Logarithmic mode on/off	Ctrl + L
Increase zoom for logarithmic mode	Ctrl + +
Decrease zoom for logarithmic mode	Ctrl + -
Increase display resolution	Ctrl + F3
Decrease display resolution	Ctrl + F2
Increase contrast	ALT + F3
Reduce contrast	ALT + F2
Increase brightness	Shift + F3
Decrease brightness	Shift + F2
Activate/deactivate harmonic mode	Ctrl + H
Switch between relative and absolute bandwidth in harmonic mode	ALT + H

Navigation and playback

Play	Space
Sets the play cursor left	Left arrow
Sets the play cursor right	Right arrow
Sets the play cursor to object end	End
Plays as loop	Ctrl + Spacebar
Plays with 7s pre-roll time in front of play cursor	NUM_1
Plays with 3s pre-roll time in front of play cursor	NUM_2
Plays with 0.3s pre-roll time in front of play cursor	NUM_3

Loads entire zoom range (X and Y axis)	NUM_0
Loads user-defined zoom range (X and Y axis)	NUM_4 to NUM_9
Loads user-defined zoom range (X axis only)	STRG + NUM_0, STRG + NUM_4 bis STRG + NUM_9
Loads user-defined zoom range (Y axis only)	ALT + NUM_0, ALT + NUM_4 bis ALT + NUM_9
Saves user-defined zoom range (X and Y axis)	STRG+ ALT + NUM_4 to CTRL + ALT + NUM_9
Loads user-defined zoom range (X axis only)	1-5
Loads user-defined zoom range including FFT block size	6-9
Saves user-defined zoom range (X axis only)	CTRL + ALT + 1 to CTRL + ALT + 5
Saves user-defined zoom range including FFT block size	CTRL + ALT + 6 to CTRL + ALT + 9
Saves only user-defined FFT block size	CTRL + ALT + SHIFT + 6 to CTRL + ALT + SHIFT + 9

Edit selection

Move selection horizontally only	ALT + LCLK
Move selection vertically only	SHIFT + LCLK
Double-clicking on the selection applies it to the entire frequency range.	DBLCLK
Copy selection (settings only)	CTRL + C
Cut selection (settings only)	CTRL + X
Add selection (settings only)	CTRL + V
Multiple selection	CTRL + LCLK

Click marker

Create Marker	SHIFT + C
Jump to next marker (all markers)	NUM_+
Jump to previous marker (all markers)	NUM_-
Jump to next marker (without click marker)	CTRL + NUM_+
Jump to previous marker (without click marker)	CTRL + NUM_-

Edit

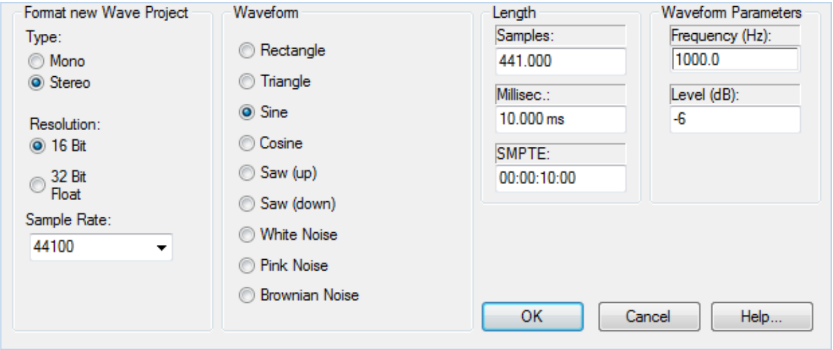
Restores standard value to an edit field	ALT + LCLK

More

Calculates	ENTER/RETURN
Solo on/off	S
Loop on/off	L
Inverse on/off	I
Opens mixer	M

Waveform generator

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



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

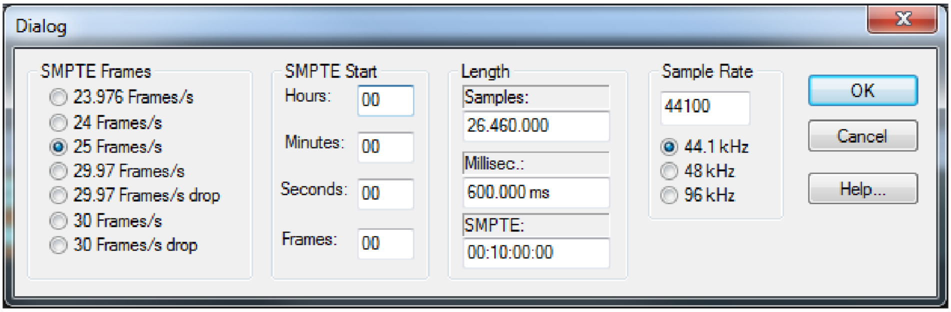
The following **sample rates** are available: 22050, 32000, 44100, 48000, 88200, 96000, 176400, 192000, and 384000.

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

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

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

SMPTE Generator



If you have no external MTC to SMPTE converter to synchronize to analog equipment, this dialog allows you to create audio files containing an SMPTE time code. To do this, place the generated SMPTE signal onto an empty track and route its output to the SMPTE sync input of the analog recording device.

You can choose from the following specifications as **SMPTE frames** per second: 23.976, 24, 25, 29.97, 29.97 drop frame, 30, 30 drop frame.

The **SMPTE start** value can be entered in hours, minutes, seconds, and frames.

Specify the **length** in samples, milliseconds, or SMPTE code.

You can choose from the following **sample rates**: 44.1kHz, 48kHz, and 96 kHz.

Process Only Left/Right Stereo Channel

Choose whether you wish to edit the left or right channel here. Of course, this is only useful if the signal you wish to edit is a stereo signal. Clicking again deselects everything.

Process Effects Offline

All effects opened using this menu are calculated destructively, provided the option "**Apply effects offline**" is active. The option to work with a copy is available in order to preserve the original audio material. The "**Create copy**" option is already selected in the corresponding dialog.

CD/DVD Menu

This menu contains special functions for the CD/DVD mastering process like setting CD tracks, sub-indices and dialogs for creating CDs or Audio DVDs.

Samplitude allows you to directly burn CDs from any project virtual project or any stereo HD wave project. A sample rate of 44.1 kHz is required. 24 bit objects are converted into 16 bit objects in the CD burning/CD trackbouncing process.

"CD arrangement" mode is especially useful if existing wave files or wave projects are going to be used to burn a CD. Wave projects are arranged behind one another on one track in the VIP when loaded. The distance is set according to the CD pause time ("CD/DVD menu -> Set pause time (view page 1007)").

Tip: If you want a finished virtual project to be burned as a track onto CD, use the command "CD/DVD -> Create CD -> Generate completely new file" (view page 986) beforehand to convert this multitrack project into a single wave project. Don't put any CD's into the drive and end the burning process after bouncing. The newly created Wave file can then be placed into the new CD VIP as a track object.

Import audio CD track(s)

This function allows the import of audio data from CD/DVD drives. The data is imported digitally which eliminates any loss in sound quality. The CD tracks are imported into the project as wave files.

To import audio tracks you should proceed as follows:

1. Open the "**CD/DVD**" menu or go to "File -> Import -> Import audio CD track(s)" (view page 573).

2. If you have more than one drive installed, click the button "CD drive options" to open the drive list dialog. Select the desired CD-ROM drive and then close the dialog by clicking "OK".

Note: The name of the currently selected CD-ROM drive is displayed above the button "CD Drive Options" along with read speed and copy mode displays.

If the drive letter of the CD/DVD burner has changed in the meantime, you will have to reset the drive list in order to guarantee correct allocation.

3. Select the desired track(s) using the keyboard combination "Shift" or "Alt" and the arrow keys.

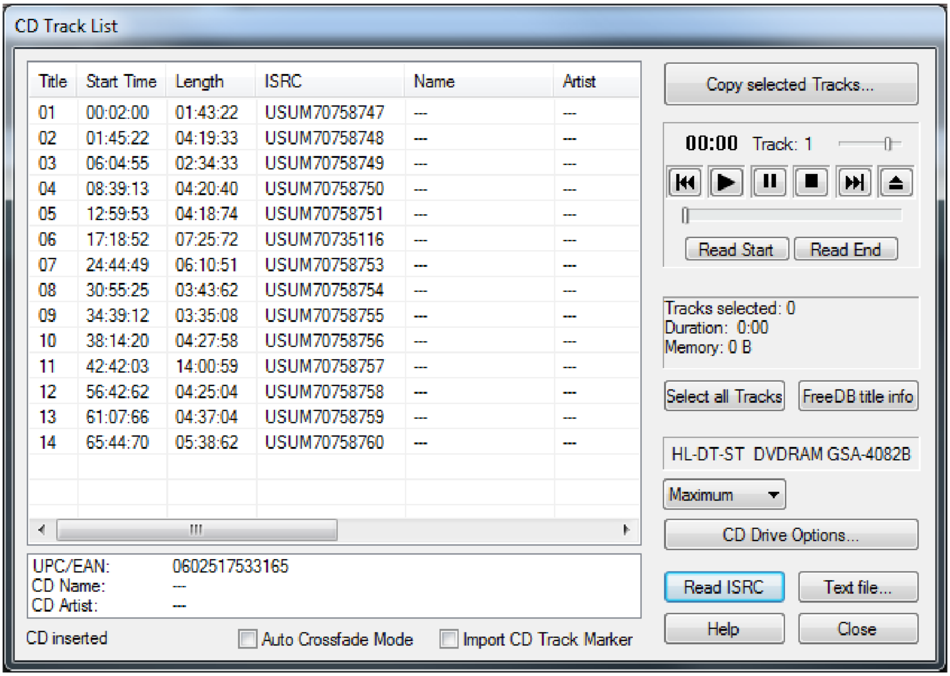
4. Click on "Copy selected CD tracks..."

5. Select a file name for the created WAV file or HD Wave Project and click on OK.

6. The audio material is now copied onto your hard drive from the CD. A progress bar indicates the status.

7. Close the track and drive lists. New objects will appear in your VIP containing the audio material from the CD.

Track List Dialog



Copy Selected CD Tracks: This button begins the audio copying process. All selected tracks are copied into a WAV file or into an HD wave project. A new object is created in the active virtual project for every track.

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

Reverse: Jumps to the previous track.

Play: Starts audio playback of the first selected track in the list.

Pause: Stops playback; start again later at the same position by pressing "Resume".

Stop: Stops playback.

Forward: Jumps to the next track.

Eject: This ejects the CD from the drive.

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

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

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

CD Drive Options: Configure the system drive list here.

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

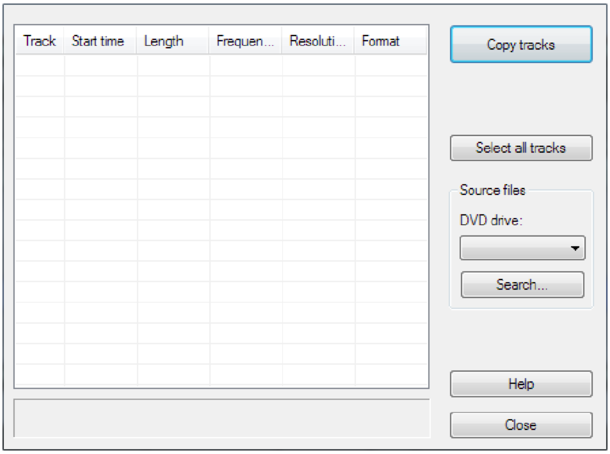
Text file: Exports all of the information listed as a text file.

Auto Crossfade Mode: Switches on Auto Crossfade Mode when importing audio tracks. Crossfades will be automatically set between imported tracks.

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

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

Import Audio DVD...



This function allows the import of audio data from DVD drives. The DVD tracks are created as wave files and integrated into the virtual project during importing.

To import audio DVD tracks into a virtual project, proceed as follows:

1. Open the dialog via "**CD/DVD**" or via "File -> Load/import -> Import audio DVD".

2. If more than one drive is installed, click the "DVD drive" list to open the drive list dialog. Select the desired DVD drive.

3. Select the desired track(s) using the keyboard combination "Shift" or "Alt" and the cursor keys.

4. Click "Copy tracks"

5. Enter a file name for the resulting wave file and click "OK".

6. The audio material will be copied onto the hard drive from the DVD. A progress bar indicates the status.

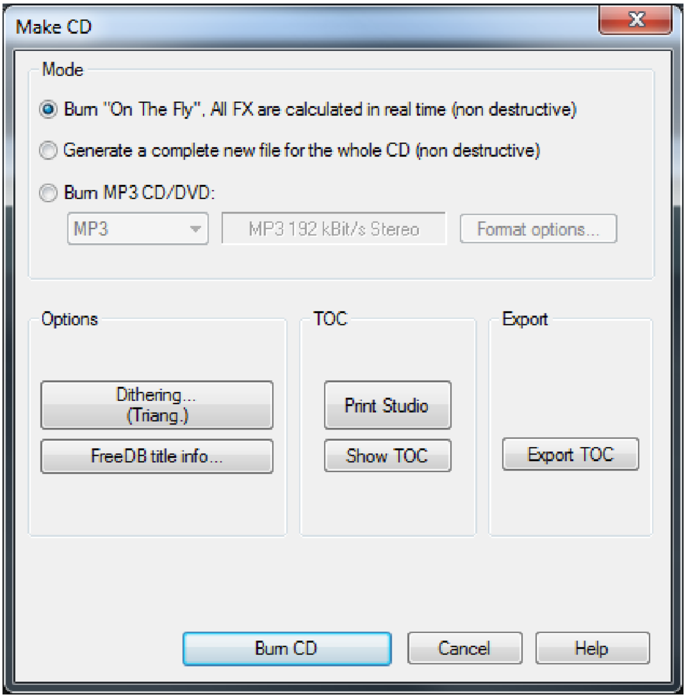
7. Close the track list. New objects will appear in your virtual project containing the selected audio material from the DVD.

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

Make CD...

This dialog starts the CD writing process. Samplitude contains high-quality and constantly updated CD burn routines which are licensed from "Point Software & Systems".

Samplitude creates a so-called "TOC file" (Table of Contents) before the CD burning process starts, and this features the name of the current project and the file extension TCX. This file is saved via the same path as the project itself.



On the fly: Use this mode if you want to write a CD directly from the project. All necessary calculations are made in real time during the burning process. This mainly involves:

- Object effects, volume, and panorama settings
- Fades/crossfades
- Mixing tracks
- Mixer track effects
- Effects from the mixer master section
- Plug-ins used in the mixer
- 32-bit float -> 16-bit conversion and dithering

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

Burn MP3 CD/DVD: All tracks are exported as individual MP3 files according to the selected format options and then loaded into the MXCDR tool. This provides the opportunity to load more files or begin burning a data CD.

Detailed information about **dithering options** is available in "Dithering options (view page 638)".

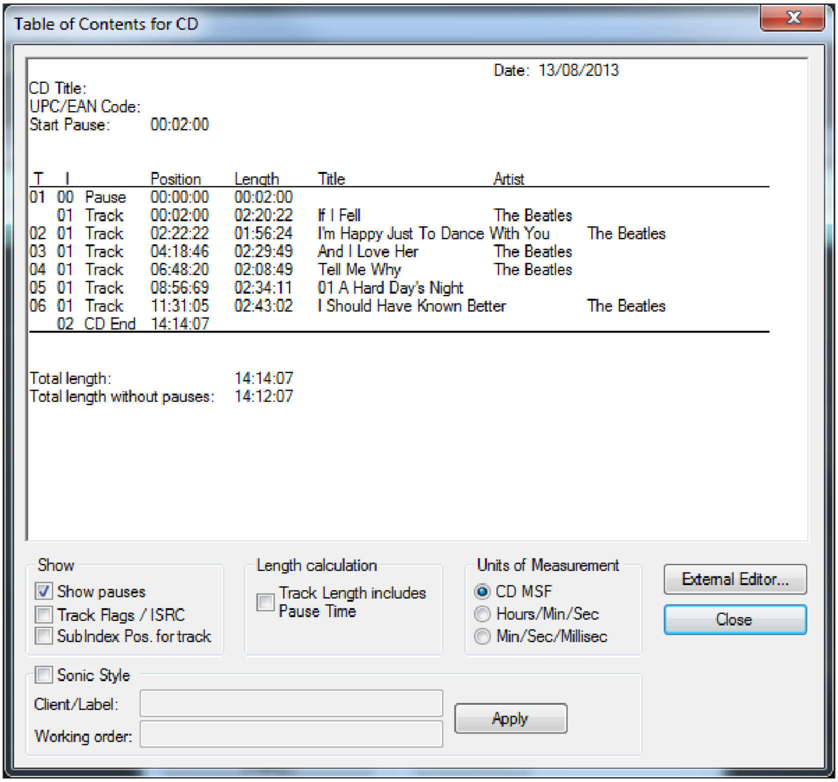
freeDB title info: This function causes Samplitude to search the freeDB.org database for information that corresponds to the track and displays it if found.

Print Studio: This button starts the external "Print Studio" program. Use this tool to easily print the content file of the current CD. Choose between a text style format to print the production documentation and a formatted printout for the CD jewel case.

Show TOC: This button opens a text window with the details of the current table of contents. You can also open these current CD details in an external editor.

Show TOC...

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



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

- Pause
- Track Flags / ISRC (view page 983)
- Sub-index position in the Track

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

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

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

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

DDP export

The additional tool DDP Export will be launched.

DDP Export is available only in Samplitude ProX Suite und Sequoia.

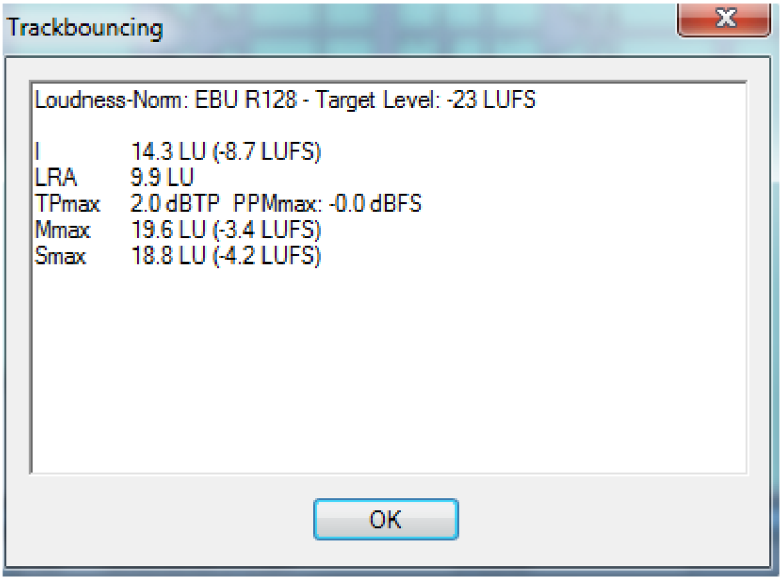
This additional program enables the export of a CD project that is compatible with DDP-CD mastering standards. It can be opened by pressing the "Export DDP" button in the "Create CD" dialog.

During exporting, the following files are created: **DDPID (DDP ID stream)**, **DDPMS (DDP map stream)**, **Image DAT**, and **PQDESC (PQ description), CHECKSUM.TXT (CRC32), and CHECKSUM.MD5**. Zip these before you send them to a CD manufacturer **via FTP**. If the manufacturer can unzip the files properly, you can be certain that the files are consistent.

The DDP ID stream (DDPID), the DDP map stream (DDPMS), the audio data (Image DAT), the PQ description (PQDESC), and the check sums CRC32 and MD5 can also be **burned to DVD** for delivery to the manufacturer with the note "Replica: CD-DA".

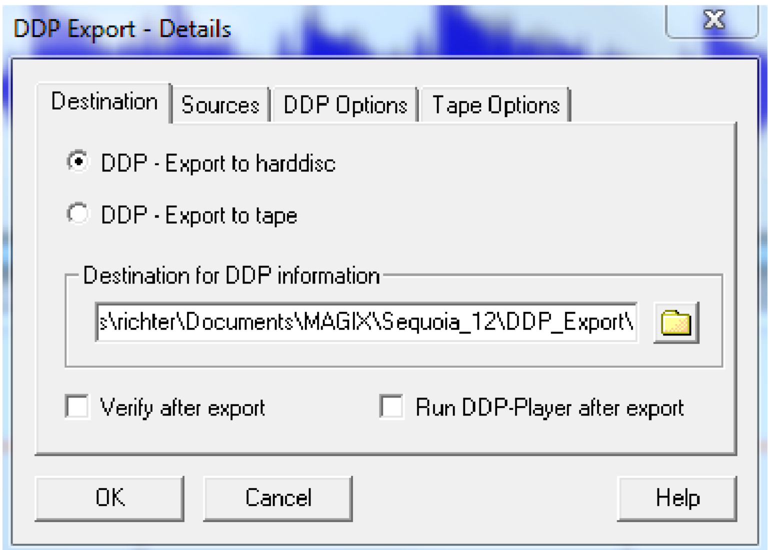
Note: Samplitude creates an additional WAV file during DDP export. This does not need to be transferred to the manufacturer, since the audio data is already included in the Image DAT file.

Simply enter a name for the project's CD audio files. Samplitude will now conduct trackbouncing of the virtual project and show the volume at the end.



At the end, the tool for DDP export will appear.

The project can be exported either to hard disk or onto tape (for example, Exabytes EXB-8505, EXB-8500). A project exported to the hard disk can be transferred to a common medium like a DVD or sent per FTP to the manufacturer.



Destination: Enter a target folder for **exporting to hard disk**. Enter the path to the tape drive in "**DDP - Export to tape**". Use the function "**Verify after export**" to start importing the CD images for verification.

The "**Run DDP player after export**" option opens the "MAGIX Sequoia DDP Player by Sonoris" directly after export.

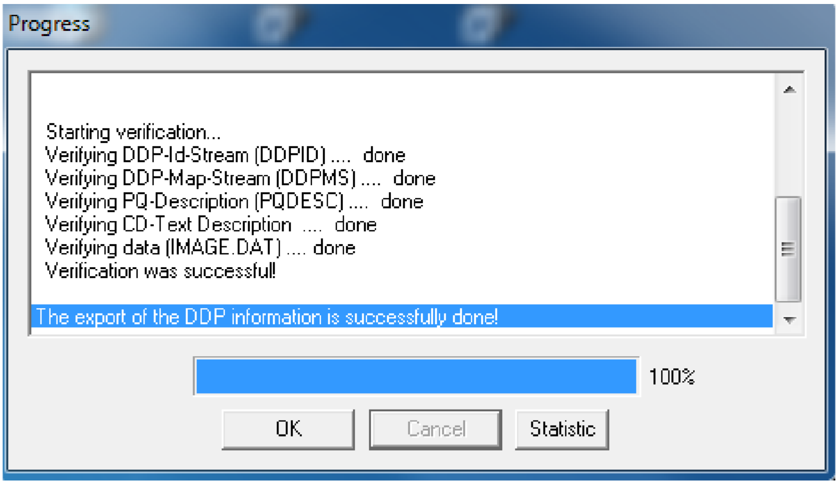
Sources: The **CD info file (*.tcd)** associated with the project is created automatically by Samplitude. If the DDP export was not initiated from within Samplitude, but rather another program outside of the Samplitude program folder, select the corresponding files here (CD info *.tcd + audio file/data file). "**CD info details**" shows the contents of the CD info file.

DDP options: The formats of the **1.01 and 2.00 standard for DDP export** can be used and a **Master ID entered**. This ID is used by the manufacturer for testing purposes.

Tape options: Enter a "**Volume ID**" and an "**Owner ID**" for archiving.

Press "**OK**" to export the CD project. Samplitude calculates and compares the check sums CRC32 and MD5 and combines them with the DDPID, DDPMS, Image DAT, and PQDESC files in the **DDP_Export** folder within the target path.

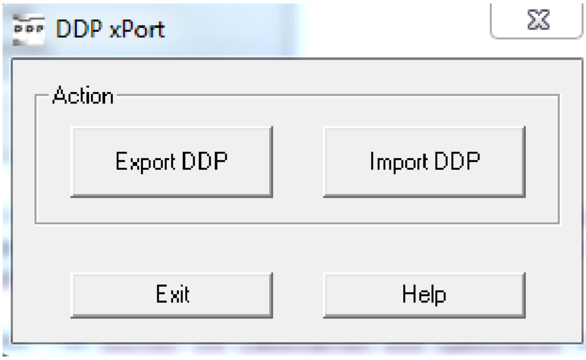
After this process is complete, you can view some statistics regarding the export process.



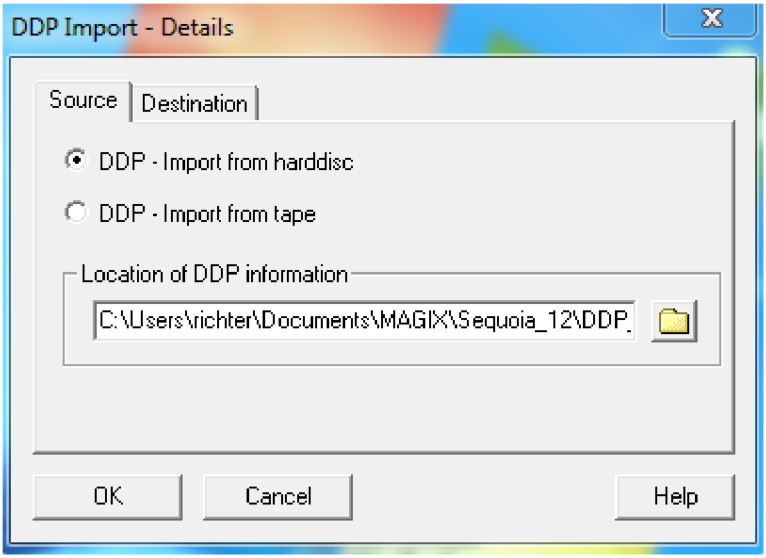
Import DDP files

To import a DDP project into Samplitude, proceed as follows:

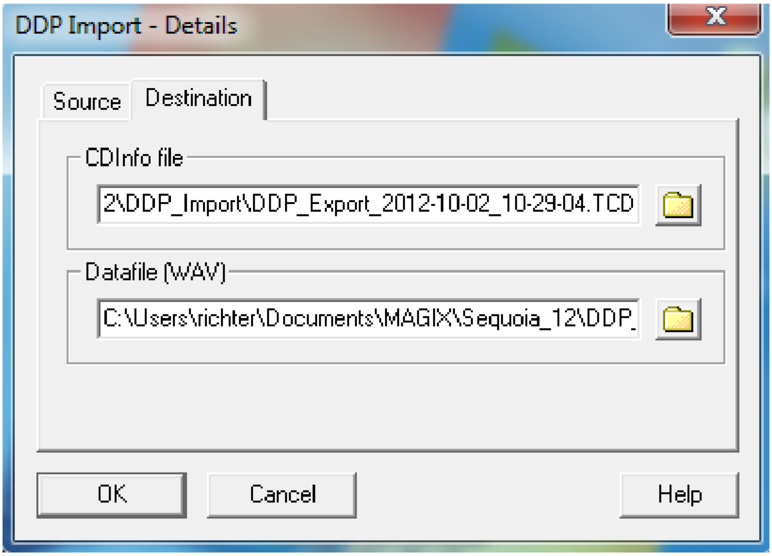
Click on the "DDP Export" button in the "Create CD" dialog or start the DDP export program from the Samplitude folder and select "Import DDP".



In the "**Source**" you can select whether you want to import from the hard disk or a tape deck. If you picked "**DDP - Import from hard disk**", indicate the folder with DDP data under "**Location of DDP information**".



The "**Destination**" tab allows you to select the file name and save location for the CD info file (*.tcd). The associated audio file will be automatically named accordingly.



Start the importing process by pressing "**OK**". The **CRC32/MD5** values are checked automatically if the CHECKSUM.TXT or CHECKSUM.MD5 are available. The result will be an audio file with the CD's entire audio data and a corresponding CD info file containing all track and other relevant CD information. All files used for this are located in the folder "**DDP_Import**" according to the target path.

A further option for importing DDP files is available through "**File > Open project > DDP import file**". A new arrangement opens including the complete CD project.

Sonoris DDP

Introduction

The Sonoris DDP Player is a standalone application that imports DDP 1.0x and 2.00 images for playback. You can playback tracks, audition gaps, see all PQ codes, ISRC, MCN and CD-Text data and automatically check the project for RedBook compatibility.

Features

- Import DDP 1.0x and 2.00 image files (loadback)
- Enhanced CD support
- Playback of tracks
- MD5-Check
- RedBook compliance checking
- Time display selectable between disc time or track time

Sonoris DDP – Basics

Audio format

A CD holds 16 bit, 44.1KHz stereo audio. This audio is stored in a whole number of frames, where every frame consists of 2352 bytes of audio. At a sample rate of 44.1KHz this means that the frame rate is 75 frames / second.

Tracks and indexes

A CD can hold 99 tracks. Normally each song gets a track number, but it is also possible to have more tracks in a song. This can be useful if you want to give sections in a song a separate track number, as is common on classical recordings.

Another, less common way to divide a track in sections is to use indexes. Some players show them but it is not widely supported. Every track can hold 99 indexes. Index 0 is a special index, it defines the pre-gap or pause before the audio starts. Most players show index 0 by a negative countdown to zero. However you don't need to use the Index 0. If you don't use it then you end up with a gapless CD (if burned in SAO or DAO mode). However, the very first index on a CD does have to be an index 0 entry of 150 frames (2 seconds) length minimum as is defined by the RedBook standard. It is also mandatory to have an index 1 for every track.

An index 0 entry normally defines a pause between two tracks, but it is possible to fill it with audio. While this is not strictly RedBook compatible, it is supported by many players.

ISRC, MCN

It is common to add a media catalog number to a disc. This MCN number the standard UPC/EAN number and should conform to the specifications of the UCC and EAN. EAN numbers are 13 digits long and UPC numbers 12. UPC numbers should therefore have an extra 0 at the beginning. The last digit is a checksum of the preceding digits.

ISRC numbers are issued per track and have the following format:

CCXXXYYNNNNN, where CC is the 2-character country code, XXX is the alphanumeric registrant code, YY are the last two digits of the year and NNNNN is a unique 5-digit number.

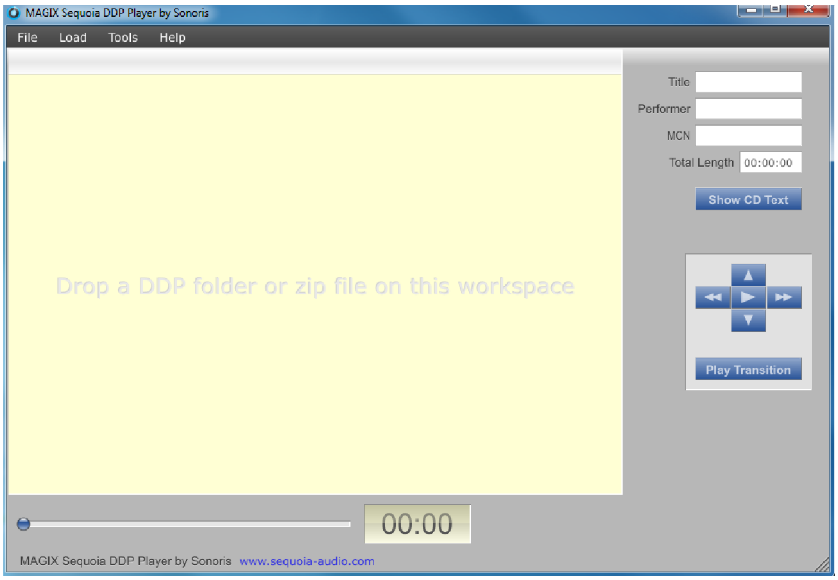
Important: There should be only one ISRC code on a track and that it is only valid on the first index entry for each track.

CD-Extra

Besides audio data a CD can also hold computer data like images, videos, song lyrics or even program files. A CD with additional data is called an Enhanced CD. It is possible to add data in the space before the first track or as a separate data session on the CD. The DDP Player supports the extra session as this is the most common and compatible format. When an Enhanced CD is played back in a normal CD player it will play just like any other audio CD. But when this CD is opened on a Windows PC or Mac it will show the data session as an extra drive.

Sonoris DDP – Main Screen

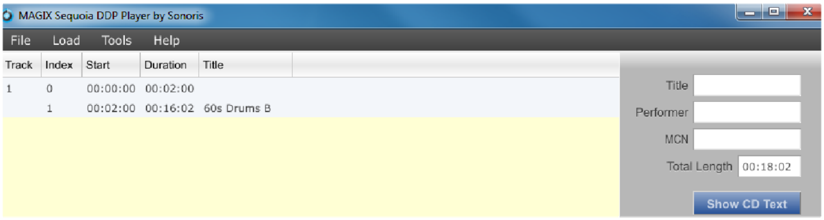
When started the "MAGIX Sequoia DDP Player by Sonoris" program interface will appear. On the left there is a large window with the project workspace. On the right there are some text boxes and a few buttons.



Workspace

The workspace consists of a table with the following columns:

- Track: track number of the entry, ranging from 0 to 99.
- Index: sub index of the entry, ranging from 0 to 99 within a track.
- Start: absolute position of the index entry.
- Duration: length of the index entry in frames.
- File Name: file that holds the audio of the index entry. This can be a wav, aif(c) or DDP image file
- Title, Performer, Songwriter, Composer, Arranger and Message: Alphanumeric information. This information is used for CD Text only .
- ISRC: ISRC code of the track.



You can adjust the columns by dragging the headers or the separators between headers. You can also hide/display columns by right-clicking on a header and selecting or deselecting the desired column.

View Disc Data

The CD Text information for the disc can be viewed by clicking on the "Show CD Text" button.

Playback tracks

To play back a track, select an index in the workspace and click the "Play" button or press the Space key. Clicking the button or pressing the Space key again pauses the playback. Playback will stop when the user presses the pause button or at the end of the project.

It is possible to navigate during playback by clicking on the left and right arrow buttons. Holding them will fast forward or rewind. The playback slider displays the relative track position and can be adjusted to go to any position quickly.

Playing a transition between two tracks

You can audition the transition between two tracks by pressing the "Play Transition" button. Clicking the button or pressing the Space key again pauses the playback. You will hear the end of the last index of the current track played into the first index of the next track and if a pre-gap exists it will be played in between. The pre-roll time can be set in the Settings menu.

Sonoris DDP – File Menu

Exit

Exits the program.

Sonoris DDP – Import

DDP Image

Loadback (import) of a DDP 1.0x or 2.00 fileset, regardless of the image data is stored in one file or more. CD Text data is imported if it exists. After the import, the project window shows the current DDP image as a regular project. You are prompted for a folder holding the DDP image.

This function enables you to import a DDP images made by any application to verify PQ points, CD text, ISRC, MCN and other data.

Sonoris DDP – Tools

Check Project

The current project is tested against a number of RedBook rules. The following criteria are used:

- MCN code must have the proper format
- First pre-gap must have a duration greater than 2 seconds
- Maximum length is 80 minutes
- Maximum number of tracks is 99
- It is also mandatory to have an index 1 for every track.
- The maximum number of index points for a track is 99
- Every track must have a duration greater than 4 seconds
- CD Text data on an index point other than 1 is not allowed
- ISRC codes must have the proper format

This test is automatically started before all export functions and PQ Sheet printing. If any of these criteria is not met an error is shown. Non critical errors can be ignored.

MD5-Check

Prompts to open a checksum file (with extension md5). It tests whether the checksums of the files found in this file matches with the actual checksums of these files. This function allows you to check the integrity of a DDP fileset after downloading or reading them from disc.

Settings

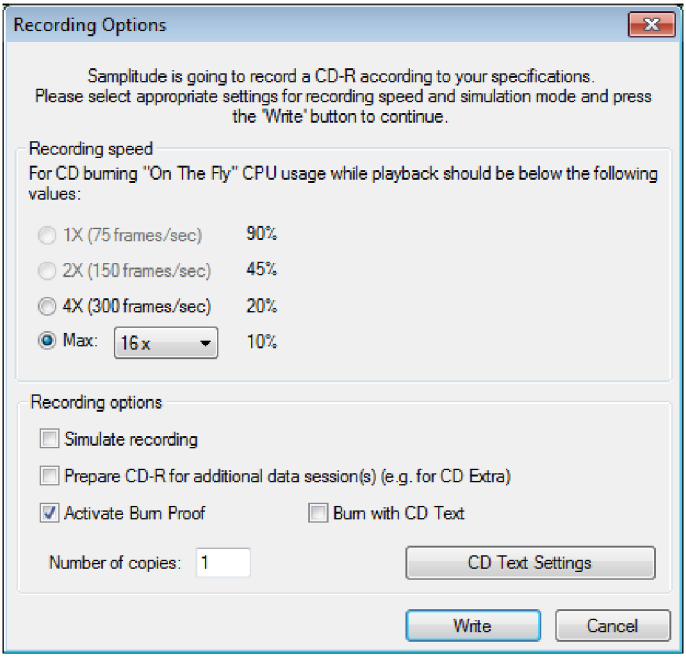
In this dialog you can fine tune the program settings. These settings are not saved with a project but will be saved as program settings.

- Pre-roll time: this setting is used during the playback of a transition between two tracks and defines how much time at the end of the first track will be played before the transition and how much time of the second track will be played into. Enter a value between 5 and 20 seconds.
- Time display: selects whether the main time display shows the track time or the disc time.

TOC Export

Clicking on the „TOC Export“ button saves the information from the CD content folder in a *.toc file (view page 988).

Burn CD



CD-R Recording Options

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

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

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

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

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

CD Text Settings: This opens the CD-Text/MP3 ID (view page 1007) editor.

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

Create Audio DVD

With Samplitude you can burn DVD audio discs with any DVD burner. You can burn DVDs in the formats +R/-R/+RW/-RW.

Note: DVD audio discs can only be played back on players with DVD audio capabilities. This is indicated by a visible DVD Audio logo somewhere on the device.

For multichannel playback your DVD audio player requires multiple analog outputs.

Samplitude burns so-called "black discs", i.e. DVD audio without graphic menus. You can create video DVDs with menus, videos and slideshows using the video editing and DVD authoring program "MAGIX Movie Edit Pro".

On a DVD audio disc you can record audio in 16-bit and 24-bit with sample rates of 44.1 and 48 kHz or even doubled and quadrupled sample rates. Up to 6 audio channels are supported in 5.1 Surround, stereo, or 4.0.

The only limitation is that the maximum guaranteed data rate of around 10 Mbit/s for hardware players cannot be exceeded. For example, for 5.1 Surround 24-bit sound, a sample rate of only up to 48 kHz is possible, since at 96 kHz, the 10 Mbit threshold would be exceeded.

The following playback times are approximations for the available space on a single layer DVD-R (44.1 kHz sample rate):

Stereo 16-bit	approx. 7 h
Stereo 24-bit	approx. 4.5 h
5.1 Surround and 24-bit	approx. 1.5 h

Create Audio DVD

First place CD track markers in your VIP. Audio CDs and DVDs can be subsequently burned from the same project.

When creating an audio DVD, the "trackbouncing" dialog will open. Select either "**Create DVD audio**" or "**DVD video with stereo PCM**".

Note: Note that the creating a "**DVD video with stereo PCM**" does not provide authoring or image information. For extensive authoring options we recommend "MAGIX Movie Edit Pro".

Format:

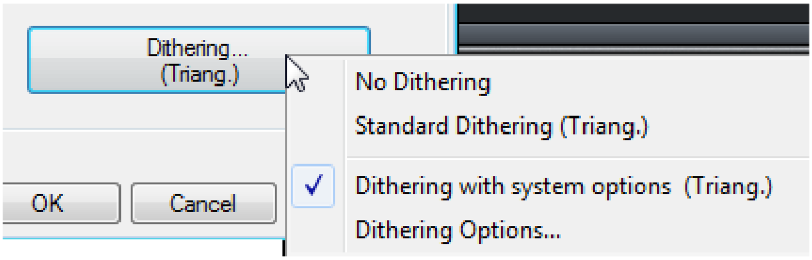
You can choose between **16 and 24-bit audio**. If "**Create DVD audio**" has been selected, then the sample rate may also be defined.

An external program will open if "**Path**" is pressed. The path for this program is set but can be changed.

Optionen:

Calculate maximum amplitude: Display the maximum volume in dB after bouncing in order to correctly set equipment for additional editing or to correct the master level. Once bouncing has been completed, the corresponding information window will be displayed.

Dithering: Every trackbouncing process may feature its own dithering process, independent of the global settings. This enables dithering to be bypassed or to apply the standard dithering (dithering featuring triangular noise).



Use this dialog to apply dithering according to the system options or to access the dithering options in the system settings. The button value in brackets (e. g. **Triang.** or **POW-r 1**) indicates the currently set dithering algorithm.

> Detailed information on this can be found in the menu reference under "File" -> "Program Preferences" -> "Dithering Options (view page 638)".

DVD-Info:

Learn details about **project length**, **maximum DVD length**, and select either **PAL DVD** or **NTSC DVD**.

Source:

Selected range only: In this case, trackbouncing is only applied to the length of the selected range in the arranger. This function does not work track-selectively, i. e. all tracks of the selected range, except for muted tracks, are used in bouncing.

From the project start to the last object: In this case, the bounce process includes all objects from the project start to the end of the last object, plus the time it takes for the sound to dissipate.

Complete project: If this option is selected, the entire virtual project is bounced.

Surround trackbouncing:

Save the individual tracks in the Surround project as mono tracks or stereo tracks.

> **Note**: Double the storage space of a DVD is required on the hard drive, since the project is bounced first and the DVD image is then created from the complete audio data that is burned.

The burn dialog provides burner selection and speed. The red "Start" button creates a DVD image first, and the writing process starts after this is completed.

Indices (Track markers)

Set CD title index

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

Use the function "Set Track Indices on Object Edges (view page 1003)" to create a track marker automatically at each object in the virtual project.

To manage the markers or rename them, use the "CD track/index manager (view page 1004)".

Shortcut: Ctrl + Alt + I

Set CD Sub Index

Choose this option to place a sub-index marker at the current playback marker position. The numbers of the subsequent sub-index markers are customized automatically.

Sub-index markers are not required for CD creation, but they allow CD players to cue to specific points within an audio track.

Set CD pause index

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

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

Set CD End Index

This command marks the end of the CD. Setting an end marker corresponds with two main applications:

1. It may be the case with some projects that the last object reverberates after its actual ending. To prevent Samplitude from cutting this section of audio during burning, you can place the CD end marker at an appropriate distance after the last object.

2. If you only want to burn a project partially to a CD, place the first track marker at the desired position in the arrangement and mark the end of the burn process with the CD end marker. Remove all track markers prior to the first track to be burned and then set the end marker after the last track to be burned.

Set Track Indices on Silence...

If you are working with a longer audio file that contains multiple tracks, this function can be used to set track markers between the titles automatically.

Min time: Period that the signal must be below the set level before a marker is set.

Threshold db: A marker is set at the positions in the arrangement where the signal sinks below the threshold volume for the set period of time.

Start number: Marker number after which automatic indexing starts.

Prefix: Enter additional characters which will precede the marker numbers set by this function. This makes it possible to easily differentiate them from markers that may already exist.

Add time (ms): Use this value to specify how many milliseconds will be set in front of the markers in the arrangement.

Delete all markers with prefix: Deletes all markers in the project featuring the set prefixes.

Delete all markers: Deletes all project markers.

Set Track Indices on Object Edges

Choose this option to place a track marker at the beginning of every object in the VIP.

Set Track Indices on Object Edges - Options

Set also Pause Indices on Object Ends

This option automatically generates pause markers at the object ends. This option allows you to set an offset for the generated track markers.

Time Offset for Indices on Object Edges

This dialog provides options for adding an offset before the object edges when placing indices.

No indices on Object Crossfades

This option prevents track indices from being placed between objects that are connected by a crossfade.

Remove Index

This function deletes a previously set track or sub-index marker. First, click the marker and access this function.

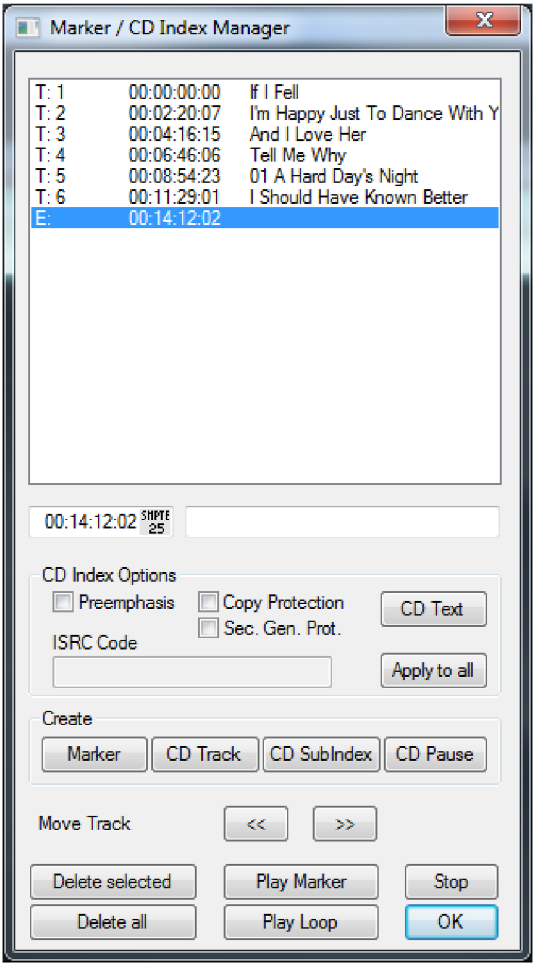
Shortcut:	Del

Remove All Indices

Choose this option to delete all previously set track and sub-index markers.

Shortcut: Ctrl + Alt + Shift + I

CD-Title-/ Index-Manager



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

CD Index Settings: Here you can set several options for the individual CD tracks, including **Copy protection (SCMS)** ,**Pre-emphasis**, **Second generation protection**, as well as **ISRC code (view page 983):**These settings can be transferred to all indices.

Pre-emphasis means that high frequencies are accentuated before A/D conversion when creating a CD's. Even during A/D conversion high frequency quantization noises will emerge. Now the Audio signal with high accentuation will be written to the CD. When playing the CD player itself will use "Emphasis Bit", which means that the frequency accentuation (highs) will be curbed. This guarantees a 'true to original' sound, likewise Pre-emphasis leads to a reduction in quantization noises.

In practice Pre-emphasis is rarely used owing to the fact that the signal noise gap in the CD creation process already large enough (16 Bit) that quantization noises are negligible anyway.

"CD text" opens a separate dialog field for entering CD text information.

You can set **new markers, CD tracks, CD sub-indices, and CD pauses** by clicking on the corresponding buttons.

Use the **double arrow buttons** to jump to the previous or next marker.

"Delete selected" Deletes the selected markers.

"Delete all Markers" Deletes all markers.

"Play marker" starts playback from the position of the selected marker.

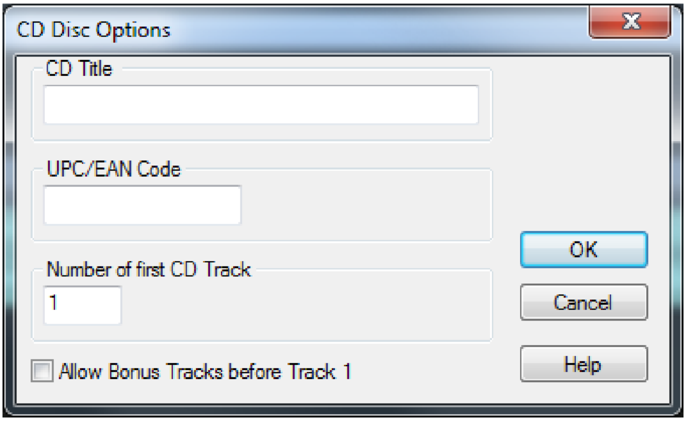
"Play loop" plays a loop around the marker.

"Stop" stops playback.

"OK" Applies your settings.

CD disc options

This dialog opens the settings for the current CD.



CD title: The title which will be written to the CD with the CD text.

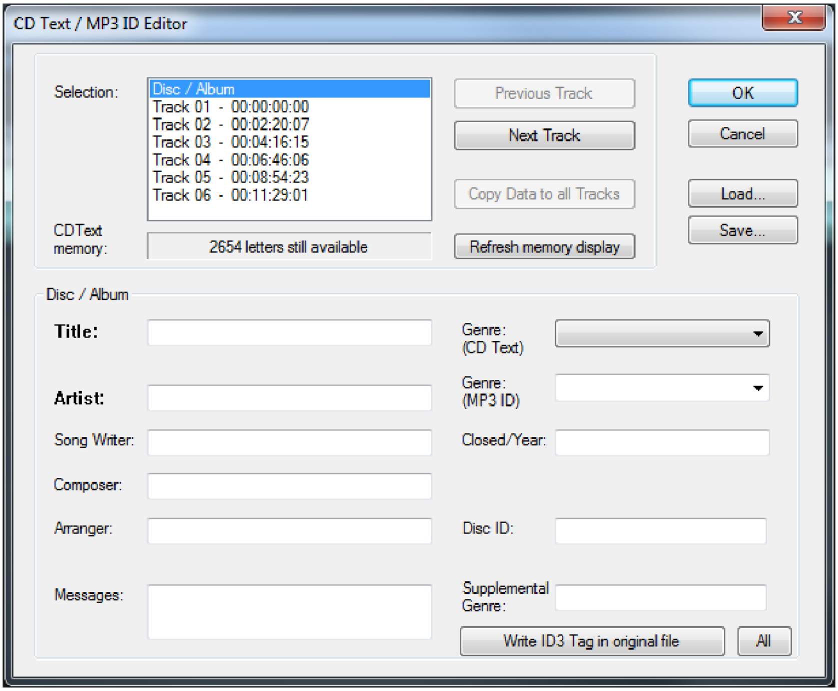
UPC/EAN Code: The EAN code (European Article Code) is a 13 digit number which is used in retail. The 12 digit UPC (Universal Product Code) is exclusively used in Canada and America and can, if necessary, be extended with a final '0'.

Number of first CD track: Under some circumstances (such as "track at once" writing), the title number of the first CD track may be set. This is not important in "disc at once" mode, since the CD will always start with title no. 1.

Allow bonus tracks before track 1: If this option is active, a so-called "hidden track" may be burned on the CD. The first track marker will now be applied to the second audio object. The "hidden track" may only be accessed on standalone CD players via the "STEP BACKWARD" button.

CD-Text/MP3-ID-Editor

Here you can enter text information for the CD you are going to burn. The CD track names are adopted from the CD track marker's labeling in the VIP. All information is also contained in the project, but in this dialog window you also have the possibility to save all entries in a separate file. (*.cdt)



Note: The number of maximum available characters for the entire CD is 2824.

If you can used MP3s in a VIP that have ID3 tags then using of the "Write ID3 tags to original file" button you can ensure that data made in the CD text/MP3 ID editor is written back to the corresponding MP3 file without separate export. This will be applied to all corresponding MP3s by pressing the "All" button.

Set Pause Time

Use this function to set the preset pause duration between objects. Audio files that you load into your virtual project will be ordered sequentially on one track. The value specified here will be set as standard pause.

Set Start Pause Time

Use this function to set the length of the pause in front of the first track. A regular value for the start pause would be about 2 seconds.

CD Arrange Mode

If this is activated, Samplitude arranges recently added objects to insert a Red Book Standard-compatible pause between the objects.

- Open a new project.
- Activate "CD arrange" mode.
- Load wave files, audio tracks, or make a recording using a microphone.

You will see gaps between the individual objects in the project. These are the inserted pauses. The duration of inserted pauses may be specified in the "Set pause duration" dialog.

Show CDR Drive Information...

Choose this option to display information about the burner.

This includes the manufacturer, drive name, hardware revision, cache size, and the features supported by the drive mechanism:

- Supports disc-at-once
- Supports indices
- Supports second generation copies
- Supports catalog numbers
- Supports ISRC
- Supports CD text
- Supports CD RW

The "disc at once" feature is extremely important for Red Book-compatible audio CD production. CDs burned in this way are accepted as masters by CD manufacturers.

Show CDR Disc Information...

This dialog displays information about the blank CD inserted into the drive, including type, track count, session count, total disk space, remaining space, and status.

The most important value is the maximum length under "Total memory" in minutes and audio frames, which cannot be exceeded during production (e.g. 359995 audio frames or 79 minutes and 59 seconds and 70 MSF).

View Menu

This menu contains tools for changing the screen view of Samplitude.

Mixer

Use this menu item to open the mixer.

Detailed information about the mixer is available in the chapter "Mixer (view page 203)".

Shortcut: M

Monitoring

This menu item opens the Monitoring section (view page 232)..

Shortcut: Shift + M

Transport Console

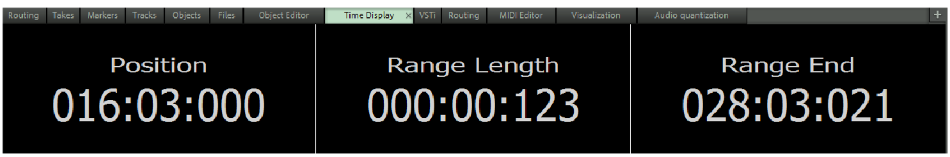
Shortcut: Ctrl + Shift + T

Detailed information about the transport control is provided in "Screen elements -> Program interface -> Overview -> Transport console (view page 90)".

Time Display

Shortcut: Ctrl + Shift + Z

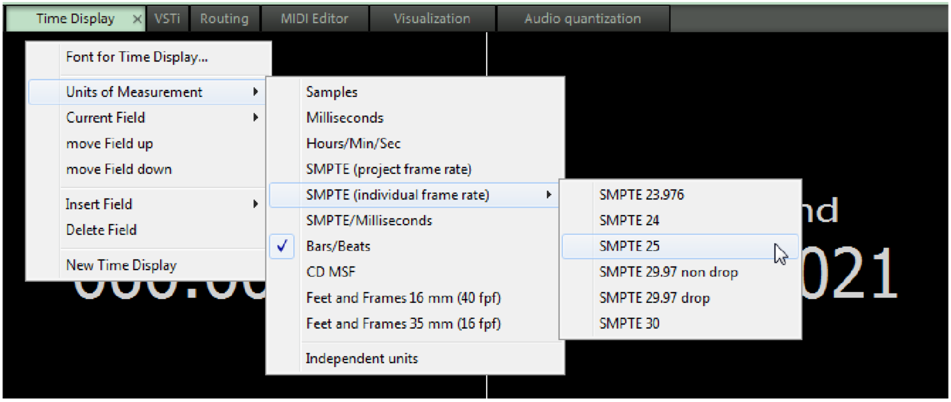
This function shows a zoom-enabled time display.



This lets you read the current time position, even if you're further away from the computer. Font and colors of the display may be selected in the context menu by right clicking on the time display.

The number of lines/fields displayed may be set between 1 and 5 in the context menu for the time display.

Right clicking lets you select and edit most sizes.



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 a + or - before the number, the change in position or range start will take place relative to the entered value.

Range length: Specify the range length here. If a new range length is entered, the range start remains the same. In contrast, if you enter a negative value, the range end will remain the same.

Range end: Display of the range end.

Object: Display of the start and end position as well as the length of the last clicked object.

Cue times: Here you can set the current playback time as well as the current remainder playback time of your cue recording (view page 725).

CD position: Here you can view the following current positions:

- Position starting at the CD start
- Position until CD end
- Position from the current CD track
- Position until CD track end
- Current CD title number

Mouse position: This value displays the current mouse position and cannot be edited.

Mixer value: This value displays the mixer fader that was just changed, as well as the volume or pan fader in the track header (cannot be edited).

Recording position: This value displays the current recording position.

Take length: This value displays the length of the current take.

Remaining recording time: This value indicates how long recording can continue until the hard disk is full.

Sync timecode input: This value shows the timecode that is applied via synchronization.

Total length: This value shoes the total length pf the project.

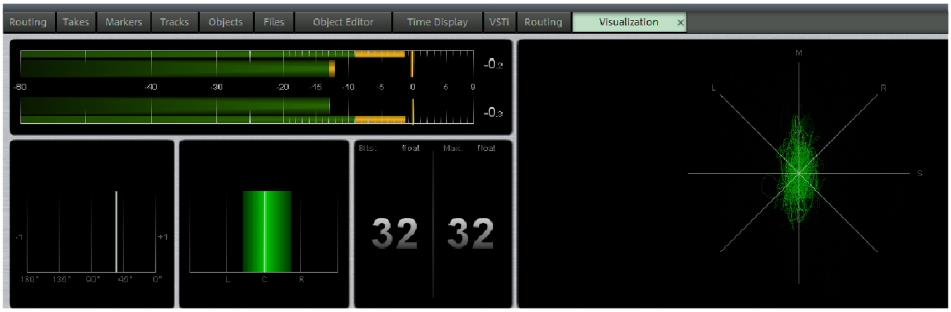
External length: This value shows the external length.

Lyrics: The content of the current and upcoming lyrics marker is displayed.

Visualization

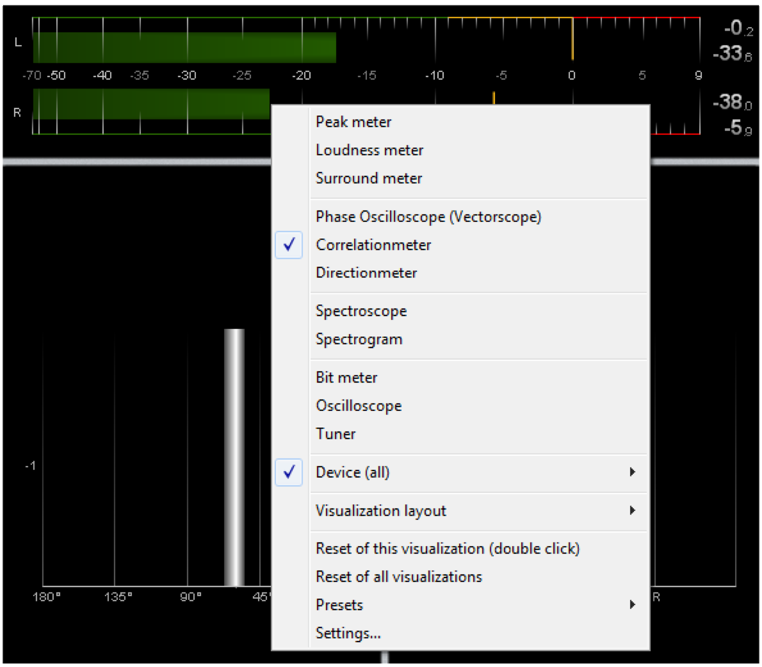
Shortcut: Ctrl + Alt + Shift + V

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



Normally the visualization is divided into several display windows. You can choose from the following display options for each window: **Peak meter** (control display), **Loudness meter (Sequoia/Samplitude Pro X4 Suite only)**, **Surround meter, Phase oscilloscope (vector scope)**, **Correlation meter**, **Direction meter, Spectroscope**, **Spectrogram, bit meter**, **Oscilloscope** and **Tuner**.

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



Device: Here you can set which output the visualization should refer to. With this option you can determine which device should be displayed if multiple audio devices are available. For example, if you mix more than four submix busses that are each routed to a different device, you can display each bus individually.

Visualization layout: Here you can load the preset **Layouts** and save your customized **Layouts**. To customize a layout, first load a layout in the format of your choice (1x1, 2x2, etc.). Then, right-click the individual sections of the layout to adjust the visualization for each.

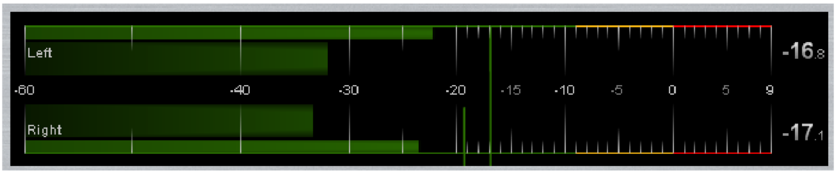
Reset this visualization (double-click): This resets the display of the current visualization.

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

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

Settings...: Click here to open the visualization settings dialog. Depending on the display instrument (which can also be changed under "Instrument" in the General tab), this dialog will show different additional tabs with settings options (see below).

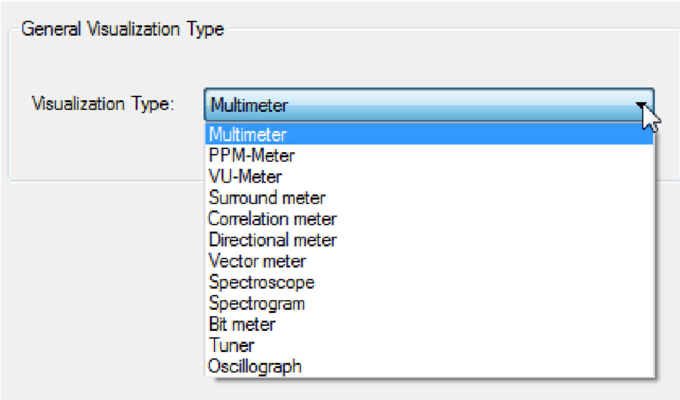
Peak Meter



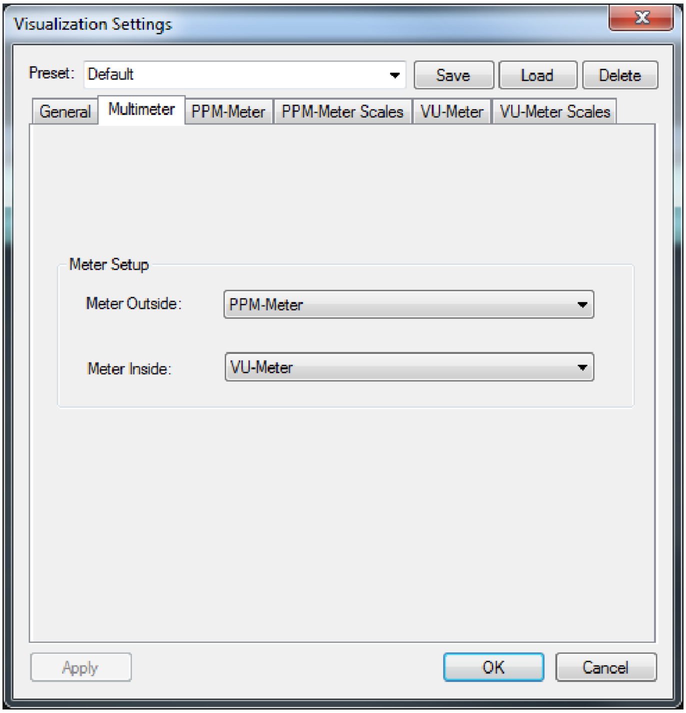
The (multi-) peak meter (Instrument multimeter) displays the volume during playback in dB. The thinner, outer bars show the **Program Peak Meter (PPM Meter)** and the thicker, inner bars display the **VU meter**, which you might be familiar with from analog equipment.

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

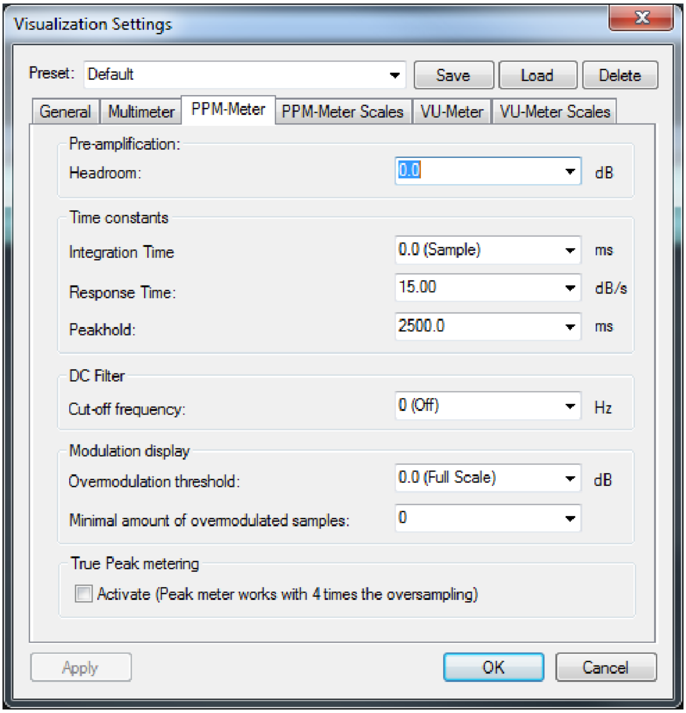
If you only want to see either the PPM display or VU display, select the corresponding display instrument.



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



PPM-Meter



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

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

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

Peak hold (ms): With this value you can specify how long the level peaks should remain. With "manual" peakhold, the level peaks remain until you double-click the visualization.

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

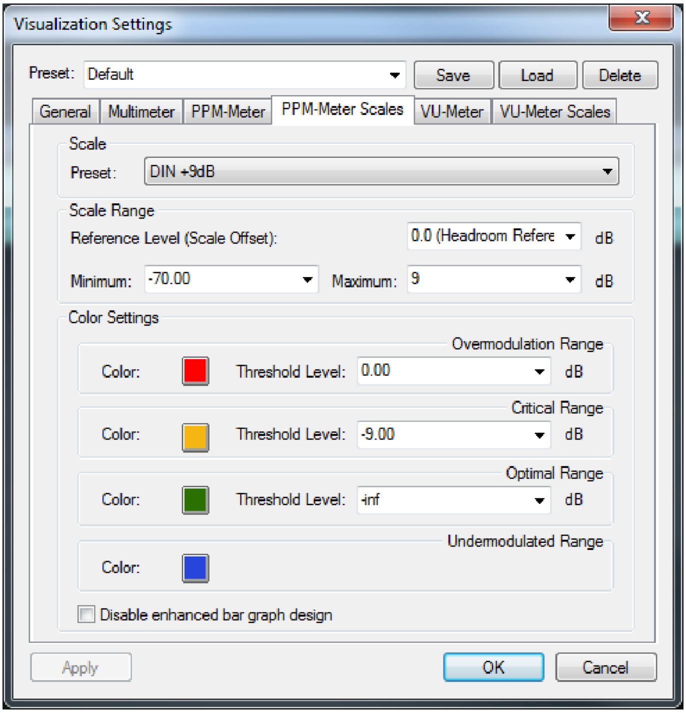
Clipping threshold (dB): This values indicates from which dB value clipping is displayed, i .e. when the display hits the red.

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

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

PPM-Meter Scale

You can choose different display options on this page.

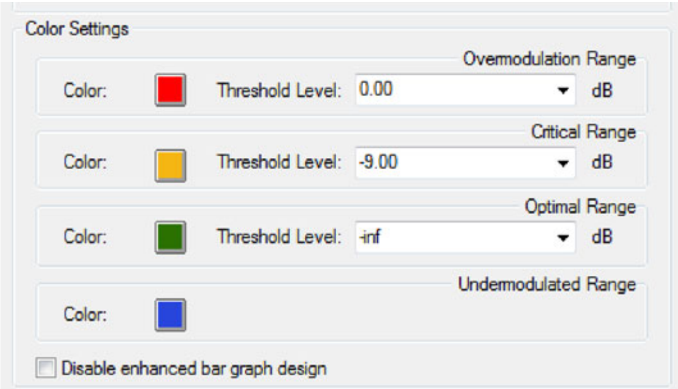


Scale: Select from a range or different scale display settings. In the "Presets" you will find a selection of peak meters with different scales and display characteristics based on the standards in various European countries.

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

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

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



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

VU Meter

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

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

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

VU Meter Scale

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

K-Metering

In the visualization area of the peak meter there are a number of presets for the K-metering system. These presets can be opened by right-clicking on the peak meter display. Here you'll find the presets „K-12 Broadcast“, „K-14 Music“ and „K-20 Cinema“. The selected scale also appears in the settings dialog of the context menu under "PPM Meter Scale > Preset".

K-metering enables uniform reference volumes for different media especially when mastering under normalized listening conditions. The peak hold display continues to show signal peaks and can be used to avoid clipping.

The "K-system" refers to the metering system developed by Bob Katz, and it has become the standard for monitoring audio signals during mastering. K-system metering enables uniform calibration and monitoring. You can use it to easily exchange audio material between different studios and have matching monitoring results. The K-metering system focuses more on musical dynamics and less on loudness. Setting the level to 0 dB sets the reference volume which no longer matches the maximum level, as often used to be the case.

Depending on the application and audio material, three different meter scales can be used:

K-20 = 0 dB reference (83 dB SPL) at -20 dB FS

K-20 is recommended when using audio with large dynamics like classical music or film sound.

K-14 = 0 dB reference (83 dB SPL) at -14 dB FS

K-14 is recommended for rock and pop productions or for Surround sound.

K-12 = 0 dB reference (83 dB SPL) at -12 dB FS

K-12 is recommended for radio and television networks.

The corresponding scale for setting up the monitoring volume is calibrated with pink noise. If you set pink noise to 0dB, you get a level of 83dB SPL, a volume reference that originated in the film world.

Loudnessmeter

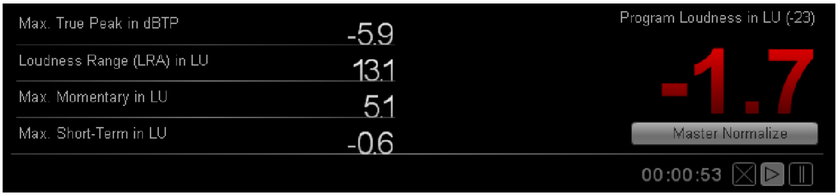
Only in Samplitude Pro X6Suite:

Loudness refers to the subjective volume sensitivity when listening to audio. This sense varies among listeners, depending on several listening conditions such as level, music style, age and even psychological state.

EBU Recommendation R128

The "EBU Recommendation R 128" deals with a method for measuring loudness which is being used more and more widely. The goal of this recommendation is to standardize all audio productions and level metering. This offers a replicable method for loudness metering. R 128 is based on the ITU-R BS.1770 standard which was established by the International Telecommunications Union.

The measurement is based predominantly on a consistent, average loudness impression. This impression is created by the integrated loudness, averaged over the entire duration of the piece. This value is then used for normalization. According to EBU R 128, the target value for normalization is "-23 LUFS".



By changing to loudness modulation and normalization, a uniform loudness level can be achieved, which can be retained for the whole production process. The end result is an even loudness impression for the listener and audio material that contains fewer artifacts caused by excessive dynamic compression.

The hope is that loudness normalization can stop the "loudness wars" and at the same time improve the audio material quality.

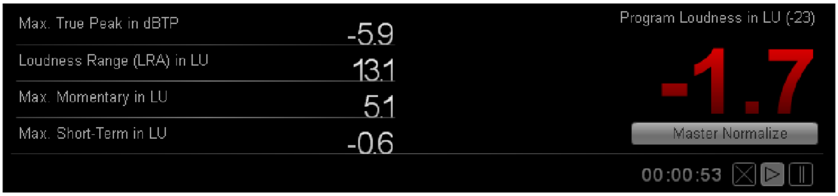
The EBU Recommendation R 128 focuses on three parameters that are essential for the properties of the audio signal:

- **Program Loudness:** Integrated long-term loudness for the duration of the audio material - measured in LUFS (Loudness Units, referenced on Full Scale) or in LU (referenced to a target value, e. g. 23 LUFS according to EBU 128). This parameter indicates the average program loudness. It calculates the mean value over the length of the piece. The term LU is the level difference to the target value. A measured calue of -20LUFS for example corresponds to a difference of 3 LU from a target value of -23LUFS. The display would show LU 3 in this case.
- **Loudness Range (LRA):** Variation in measuring the loudness of the audio material - measured in LU (Loudness Units). The value indicates the difference between the loudest and softest measured value (except silence).
- **Maximum true peak:** The maximum value of the audio signal during continual measurement on the timeline. When converting digital to analog signals level peaks could happen that were not displayed before. Thats why the displayed peak level of a True Peak Meter is typically higher than that of a usual digital (QPPM - Quasi Peak Program Meter). The R128 standard alows a maximum True Peak value of -1dBTP.

The combination of these parameters allow more dynamic mixes without affecting the loudness.

Loudness Meter Compact R128

Only in Samplitude Pro X6 Suite



Maximum True Peak in dBTP: Peak level of the audio signal, continuously measured on the time line in dBTP.

Program Loudness in LU: Relative program loudness value for the „Integrated Loudness“ in measured loudness units (LU).

Loudness Range in LU: Loudness range measured in LU as a relative value. Below are the loudness values for the "Momentary" and "Short" measurements according to the settings.

Measurement Duration: Time entry for the current duration of the total loudness measurement from Program Loudness and Loudness Range. With the three buttons at the aide you can interrupt (Pause button), continue (Play button) or reset (Reset button) the measurement. By clicking the "Reset" button all display values are reset.

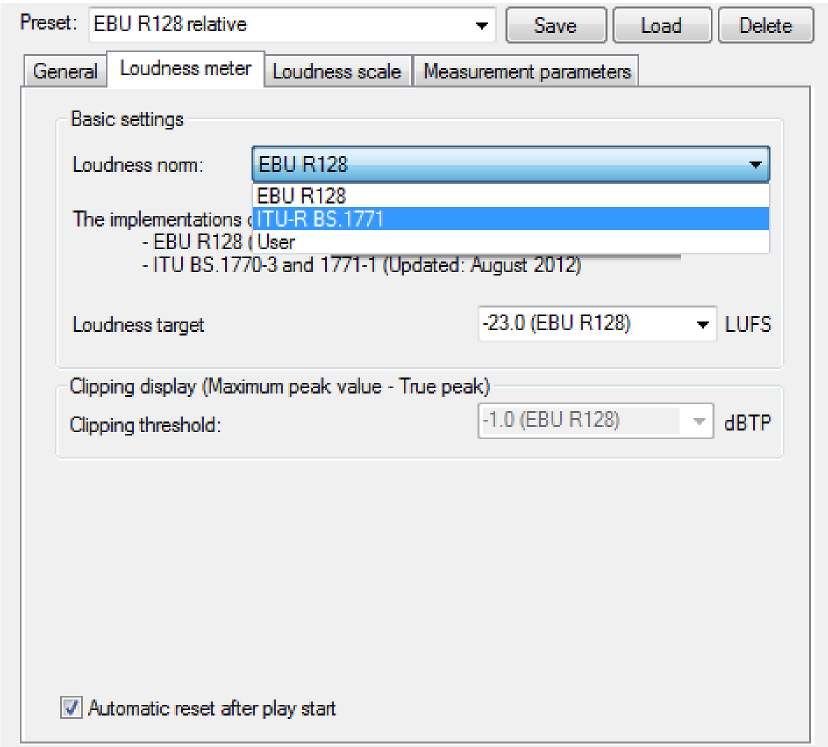
The "Master Normalization" button makes it possible to normalize the master fader at any time.

Loudness Meter - Settings

Only in Samplitude Pro X6 Suite:

You can access the loudness meter settings dialog by right-clicking on the graphic display ("Settings..."). In the **„General“** tab go to "Instrument" > "Loudness Meter" to work with the entire instrument section. The "Multimeter", "VU Meter" and "PPM Meter", in contrast, produce only a simple bar display similar to that of a peak meter. A range of presets is available in the preset list. Confirm your selection with "Apply".

In the „**Loudness Meter**“ tab you can set the standard that should be used under "Loudness Norm". If you select "User" as loudness norm you can set the parameters yourself. If you select "EBU 128" or "ITU-R BS. 1771", the parameters will be set according to the corresponding standard. Windows which are no longer available for editing will be displayed grayed out.

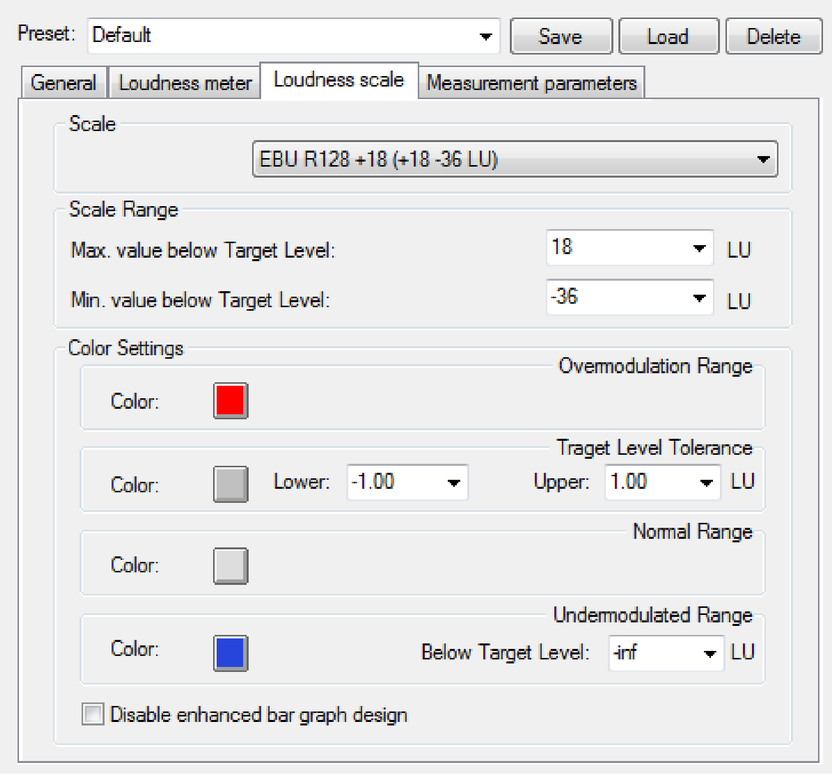


Here you'll also find the important parameter "Loudness Target Value". The target value "-23 LUFS" is specified for EBU R128. This level represents the default value for content production in broadcasting.

Clipping Threshold: Set the clipping level, i.e. from what point clipping should be indicated. This value is standardized for EBU R128 at -1 dB. This results in a measured True Peak-peak level of -0.8 dBTP being displayed as clipping for example.

Automatic reset at playback start: With this option you can reset the measurement each time you begin playback. This option represents the "Reset" button in the graphic display.

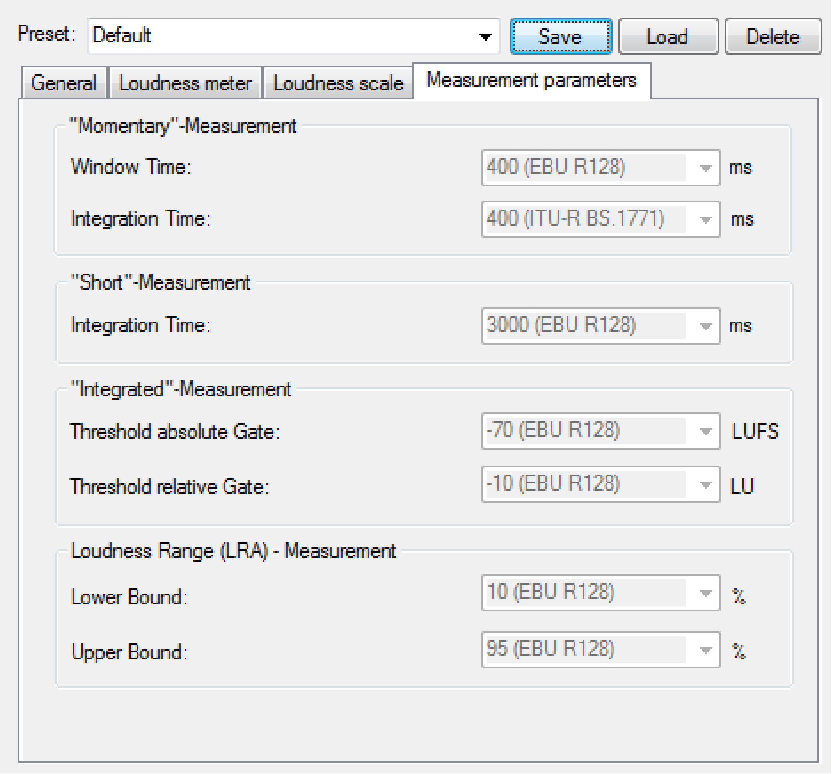
In the "**Loudness Scale**" tab you can switch between different standardized scale norms. The range of values can be defined freely. The entry "EBU R128 +9" means that the scaling display is sufficient until +9 LU is reached.



In the scale area you can determine by how many LUs the maximum or minimal value is allowed to exceed or fall short of the target value.

Under "Color Settings", you can define color coding as well as thresholds for the clipping section, tolerance section around the target value, the normal value and the weak signal section.

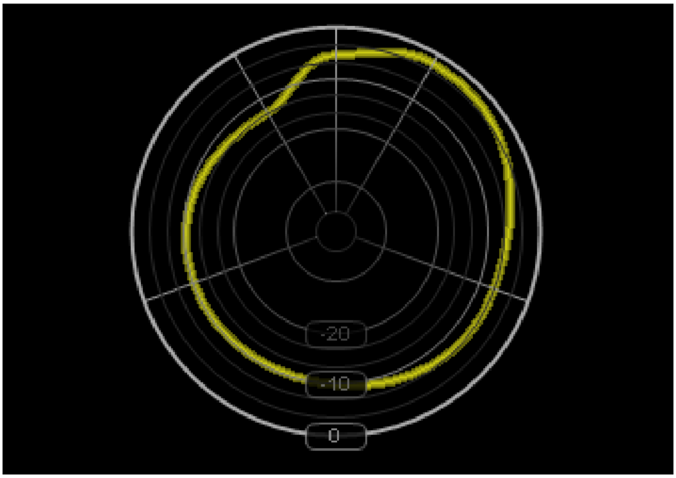
In the "**Measurement Parameters**" tab you can set the time parameter for "Momentary" and "Short" measuring, the gate threshold for "integrated" measuring as well as the upper and lower limits for the "Loudness Range (LRA)" measuring. The specified default values for EBU R128 are marked in a noticeable way.



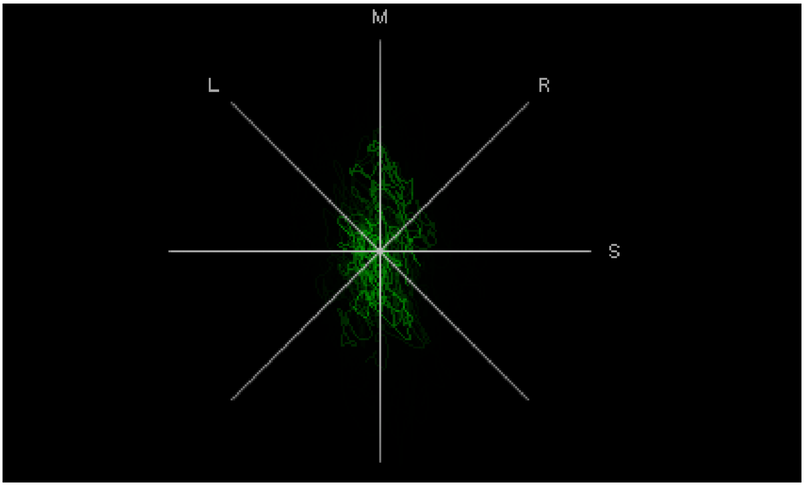
In this section when changing the norms you can adjust the affected values and save them as new presets. Absolute and relative values are displayed in relation to the selected scale. L or LK denote relative scales, LUFS and LKFS denote absolute scales.

Surround Meter

The surround meter displays the signal using the surround settings that are set as global in the „Surround Setup" (view page 298) dialog. It shows each speaker's level in a polygon on a sound representative area.



L/R Oscilloscope (phase correlation):

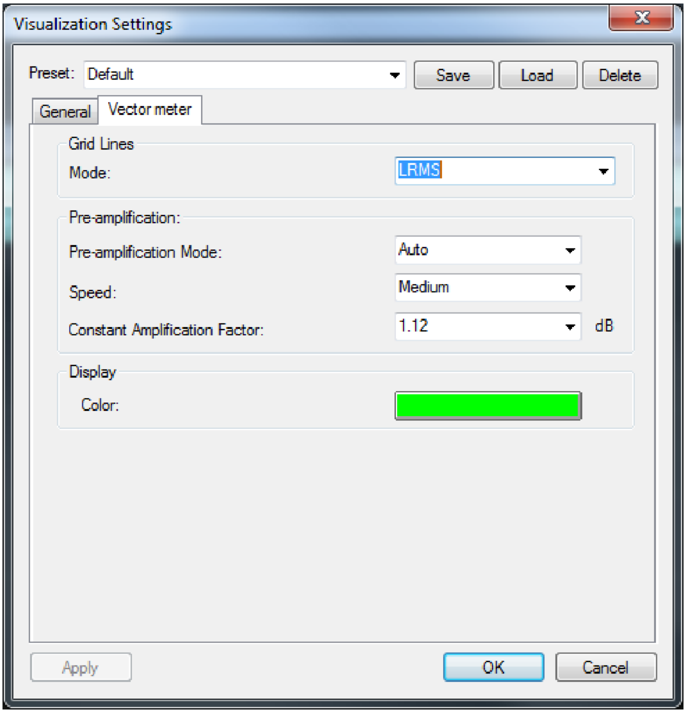


Phase correlation provides information on the distribution of the stereo image in your recording. A mono recording would be displayed as a vertical bar in this view. A song produced in stereo is by contrast shown as "diffuse ball," because multiple instruments were assigned to the mix in different panoramic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

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

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

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

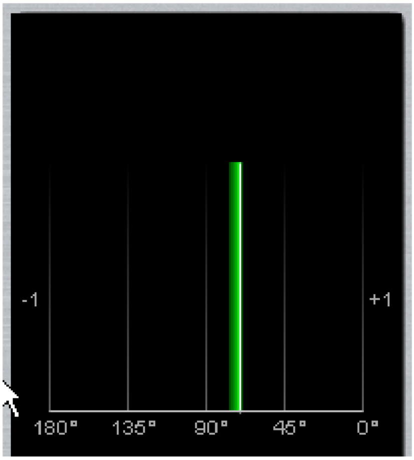


The image is correspondingly increased in pre-amplification mode, to ensure readability. You can also switch off this mode and set a constant amplification factor instead.

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

Finally you can set the display color.

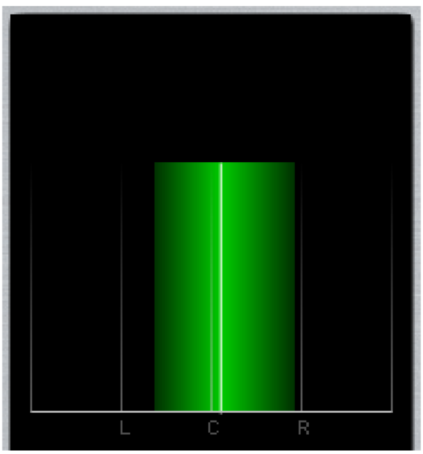
Correlation meter



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

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

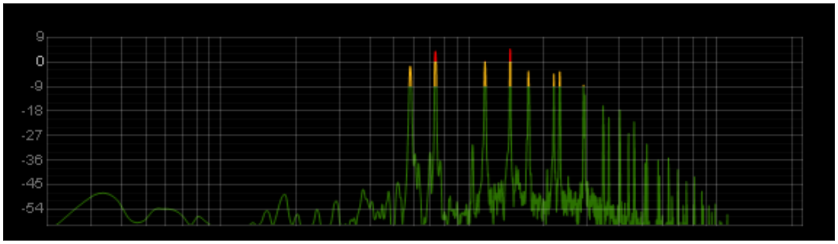
Direction meter



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

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

Spectroscope



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

Spectroscope settings

Line/bar representation: Select the representation type you want to use. **Line (smoothed)** is set by default. Other options include **Line** (connects the points with straight lines instead of interpolated curves) and Bar, where you select the number of bars.

Stereo Options: Left+Right is set as the default (i.e. mono sum). It is also possible to set up two spectroscopes, one each representing the left and right signals. The side signal can also be represented by selecting Left-Right.

FFT parameters

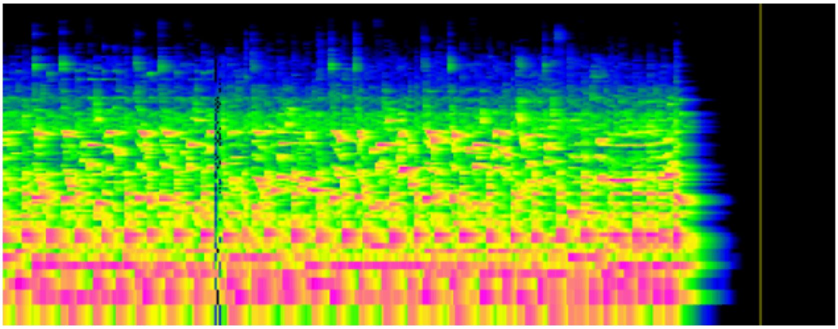
Here you can adjust the parameters **FFT Size**, **Overlap** and **Window** for determining the spectral values.

Time constants

The time constants correspond to those of the peak meter (PPM). (However, the peakhold value has no meaning for the spectrogram).

The **Spectroscope Scales** settings correspond to those of the peak meter (view page 1013).

Spectrogram

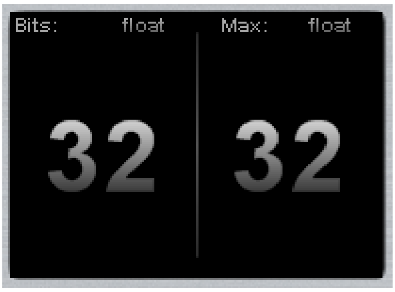


In the Spectrogram the signal is displayed as frequency proportions in a time curve. The volume of frequencies is visualized by its brightness.

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

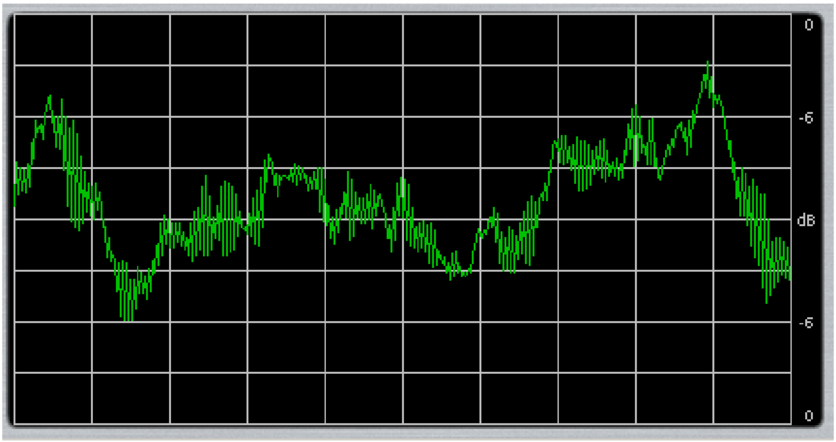
In Samplitude you can remove unwanted noise with the help of Spectral cleaning (view page 111).

Bit Meter



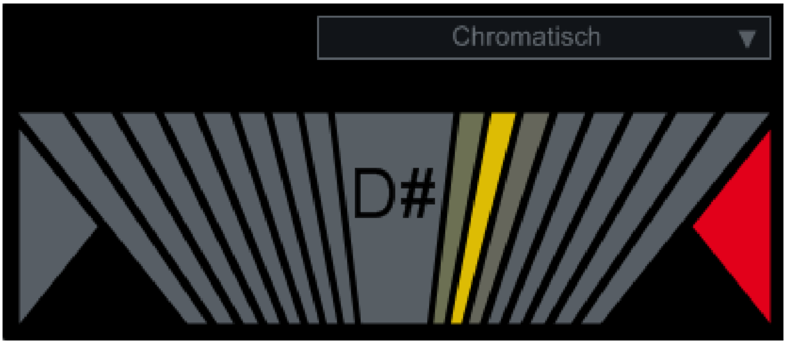
The bit meter shows you the rate at which the signal is being calculated at and which maximum editing rate is possible.

Oscilloscope:



The oscilloscope displays the signal amplitude over the time curve.

Tuner



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

This is how you can tune your guitar:

Step 1: Select your guitar in

Step 2 Open recording options by right-clicking on the record button in the transport console.

Step 3: Activate the "Monitor" button and use the monitoring mode on "Track FX Monitoring" or "Mixer FX Monitoring" to be able to control tuning acoustically.

Step 4: Activate the "Visualizer" button and select the "Tuner" visualization in the graphic display.

In the middle you will see the note name of the string being played. The calibration marks show deviation of played to exact pitch. Red triangles to the right and left show the direction the strings should be tuned. If the tuning is exact, the triangles turn green.

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

Video Window

This function opens the video window, e.g. to apply a media link.

Track Editor

This menu item opens the track editor on the left edge of the arrangement window. This enables access to all important track parameters for the selected track.

> Detailed information about the track editor is provided in "Screen elements -> Track editor"

Shortcut: Ctrl + Alt + Shift + E

Keyboard

Use the keyboard (view page 200) to access an on-screen keyboard to play MAGIX synth sounds and record them directly.

Manager/Docker

You can open the Manager/Docker here. In addition to the Manager windows (view page 174) now Visualization, Time display, Transport console, Sound pool, Object Editor and MIDI Editor are grouped as a tab window by default in a "Docker".

Manager

The manager features the following sub-windows:

- File Browser (Ctrl + Shift + B)
- Object Manager (Ctrl + Shift + O)
- Track Manager (Ctrl + Shift + S)
- Marker Manager (Ctrl + Shift + Alt + M)
- Range Manager (Ctrl + Shift + Alt + B)
- Take Manager (Ctrl + Shift + Alt + T)
- VST Instruments Manager (Ctrl + Shift + I)
- Routing Manager (Ctrl + Alt + Shift + R)
- Soundpool Manager

Detailed information about the individual manager sub-windows is provided in the "Manager (view page 174)" chapter.

Control bars

Main Toolbar

Use this command to hide or show the toolbar. A check behind the menu item indicates that the toolbar is visible.

Main Toolbar 2

Use this command to hide or show the second toolbar. A check behind the menu item indicates that the second toolbar is visible.

Statusbar

Use this command to hide or show the status bar. A check behind the menu item indicates that the status bar is visible.

Autoscroll

Switches Autoscroll on/off

See "Playback -> Playback Options (view page 727)".

Soft Autoscroll

Toggles between page and soft autoscrolling.

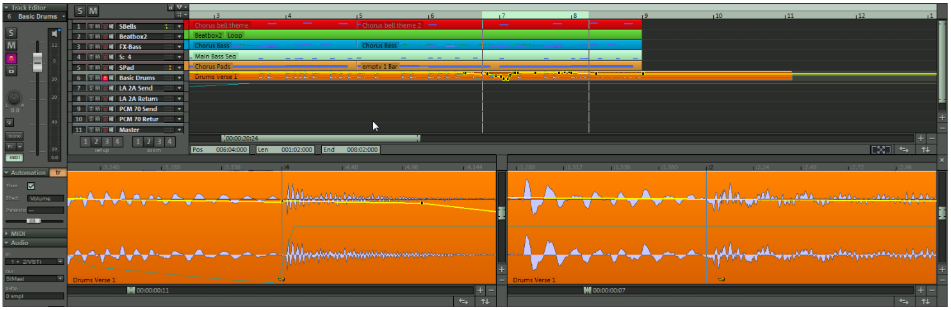
See "Playback -> Playback Options (view page 727)".

Rebuild Graphic Data

The graphical display of a project is recalculated in this case. This is usually unnecessary, but does help with errors or inaccuracies in the display after offline editing.

Sections

These Menu items allow you to take advantage of Samplitude's ability to simultaneously display **one two or three** different "views" of the same project.



If you select the option "**2 (sections)**", two view windows, each of which can be worked in independently, open **one under the other**. This way, for example, you can view the complete sample in one window, while magnifying a specific area in the other.

The "**3 (sections)**" mode can be especially helpful when searching for loop points in the wave window. In the **top large window the entire sample** can be viewed, in the **bottom left window area the beginning** can be viewed and **in the bottom right window area the end of the loop** is displayed. This view can also bee seen by using the shortcut "B". The corner values of the three windows can be independently adjusted.

You can determine the areas through window borders once and for all by selecting a start of a segment by clicking the clip bar at the corresponding position, and then selecting the end of the segment in a different clip with the Shift key held down.

When using the zoom function on a specific clip, it is necessary to first select it by using scroll bars to the right or to the bottom.

Shortcuts:

Show range:	Shift + B
Split range (shows 3 sections)	B

Activate destination

This command activates the previous/upper section.

Shortcut:	Page up

Activate source

This command activates the next/lower section.

Shortcut:	Page down

Hide Submix/AUX Busses

Use this command to remove the buses from view in the arranger window. This may improve the display in case you are working with numerous tricks.

> **Attention**: This function keeps bus tracks hidden, even if a check is set in the track manager next to the "Arrangement" column.

Grid view

Grid

This command activates the coordinate grid for the project window.

Grid Lines

This provides a choice between various line layouts for the grid display. The display of the grid helps makes the positions in the arranger window more legible.

Units of Measurement

Specify the units of measurement for the grid view here.

The following units are available: samples, milliseconds, hours/mins/secs, SMPTE, SMPTE/milliseconds, CD-MSF, feet and frames 16 mm (40 fpf) and feet and frames 35 mm (16 fpf), noise reduction (meter).

The units of measurement selection will affect the start and length of the current range in the project window, the grid width and the details of the position of the playback marker (position line).

Snap active

This functions turns the grid on or off.

Snap/Grid

Here you can set the grid type. You can choose from:

- Objects
- Range
- Beats/Bars
- Bars (relative)
- Grid/Frames
- Frames (relative)

Detailed information on snap and grid settings can be found under "File > Project Properties > Snap and Grid Setup (view page 595)".

Snap and Grid Setup

Detailed information on snap and grid settings can be found under "File -> Project Properties -> Snap and Grid Setup (view page 595)".

Show 2nd Grid Line

This command displays a second grid.

Exchange Grids

This command exchanges the upper and lower grids.

VIP Display Mode

- Definition...
- Mode 1
- Mode 2
- Toggle between mode

Detailed information on display options can be found in "File > Program Properties > Display Options (view page 631)".

- Spectral Display
- Waveform display

Representation of the waveform as Spectrum or WaveColor Detailed information on display options can be found in "File > Program Properties > Display Options > Waveform Color".

Overview Mode

Detailed information about Overview Mode can be found in the chapter „Working in the Project Window“ > „Zoom“ > „Zooming with Overview Mode“ (view page 125).

Zoom Into Selected Objects

This command displays the selected object at a zoom level that optimally fills the section. This option is also available as a button.

Horizontal

This menu contain horizontal functions for clip control, which are also partially represented in the bottom position bar as buttons.

Shortcuts:

Half section left	Ctrl + Alt + Left arrow
Half section right	Ctrl + Alt + Right arrow
Cut to playback marker	Ctrl + Alt + ,
Clip to the start of field	Ctrl + Alt + B
Clip to end of field	Ctrl + Alt + N
Zoom in	Ctrl + Left arrow, Up arrow
Zoom out	Ctrl + Right arrow, Down arrow
Show all	Ctrl + Alt + Up arrow
Zoom to range	Ctrl + Alt + Down arrow

If you can't see the playback marker in the current VIP's image section, then the command "Section to play cursorplayback markerposition" moves the visible section to the current playback marker position. In a stopped state the picture section jumps to the previous playback position.

Vertical

This menu contains vertical functions for clip control, which are also partially represented in the bottom position bar as buttons.

Shortcuts:

Half section up	Shift + Page down
Half section down	Shift + Page down
Zoom into waveform	Ctrl + Page up
Zoom out of waveform	Ctrl + Page down
Zoom Out Volume Automation	Ctrl + Shift + Page up

Zoom Out Volume Automation Ctrl + Shift + Page down

Tile

This function arranges all open windows one one top of the other.

Shortcut: Enter

Untile

This function arranges widows in the same way as they were prior to "tile vertically".

Shortcut: Shift + Enter

Window Menu

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

Close all windows

This closes the currently opened project. A confirmation will ask if you want to save the project.

Shortcut: Ctrl + H

Show/Hide Effect Window

With this function you can minimize an open effect dialog window or maximize a hidden effect dialog window.

All windows on main screen

If you have edited a project on a notebook with a second monitor connected, and you open the project later without this monitor, it can happen that windows are not accessible because they are on the second monitor. You could solve this problem by loading one of the saved default workspaces, but in doing so you would lose your layout customizations, i.e. the open windows and their docking status. This command simply moves all open windows and docks to the first monitor.

Set focus on project window

This commands activates the project window.

Shortcut: Ctrl + P

Activate docking window

This command activates the docking window.

Shortcut: Ctrl + B

Close current docker tab

This command closes the currently opened docker tab.

Keyboard shortcut: Ctrl + F4

Next docker tab

This command moves you from one docker tab to the next and activates it.

Minimize docker

This command minimizes the docker.

More...

Select Workspace/Mixer snapshot/Mixer View snapshot

With these commands it is possible to load Workspaces, mixer snapshots or mixer view snapshots. Under these menu items you will find submenus with the available workspaces and snapshots, from which you can select the desired entry with the arrow keys and confirm with the Enter key. If you also assign a keyboard shortcut to the menu items, you can switch workspaces and snapshots completely by keyboard control.

Store Position and Zoom Level

Save the current position and zoom level here. These may be retrieved via the menu item "Get position and zoom level" in the "View" menu and on the number pad of your keyboard.

The first three numbers on the keypad may be used to save presets. These configure the setup buttons 1, 2, and 3 in the arranger window.

Shortcut: Ctrl + Num 1

Ctrl + Num 2

Ctrl + Num 3

Get Position And Zoom Level

Load the saved position and zoom level here. This may be retrieved via the number pad on the keyboard.

The first three numbers on the number pad can be used to save presets.

Shortcut: Num 1

Num 2

Num 3

Store Zoom Level

Save the current position and zoom level here. These may be accessed via the menu item "Get position and zoom level" in the "View" menu and via the number pad.

The numbers 4, 5, 6 on the keypad may be used to save presets. These configure the zoom buttons 1, 2, and 3 in the arranger window.

Shortcut: Ctrl + Num 4

Ctrl + Num 5

Ctrl + Num 6

Get Zoom Level

Load the zoom level here. This may be retrieved via the the number pad on the keyboard.

The numbers 4, 5, and 6 on the keypad may be used to save presets.

Shortcut: Num 4

Num 5

Num 6

System Information

Use this function to open a new window with the system information.

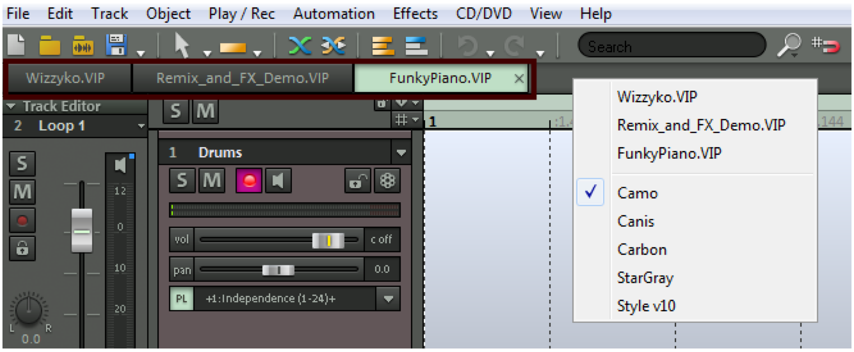
Display the date/time, number of open files, the free memory on all connected hard drives, the configuration of Windows resources, and the amount of memory used by Samplitude.

FTP Download...

This menu item opens an FTP download window. Enter your FTP access data and click on "Connect..." to establish an FTP connection immediately. Files may be downloaded, but not uploaded.

Activating a Project

Samplitude shows a list of the currently opened projects in the last section of the "Window" menu. The active project is indicated by a check mark. Select a project from this list to activate it in the project window. All of the open projects appear as project tabs in the arranger.



When you click on one of these project tabs, Samplitude will open the desired project.

Help Menu

Contents and Index

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

Shortcut key: F1

Help Index...

Choose this option to display a list of available help contents.

Context Help...

This command turns the mouse cursor into an arrow with a question mark. Click any menu item or button in one of the toolbars to receive more information.

Shortcut: Shift + F1

Manual

Use this command to open the Samplitude manual (PDF document).

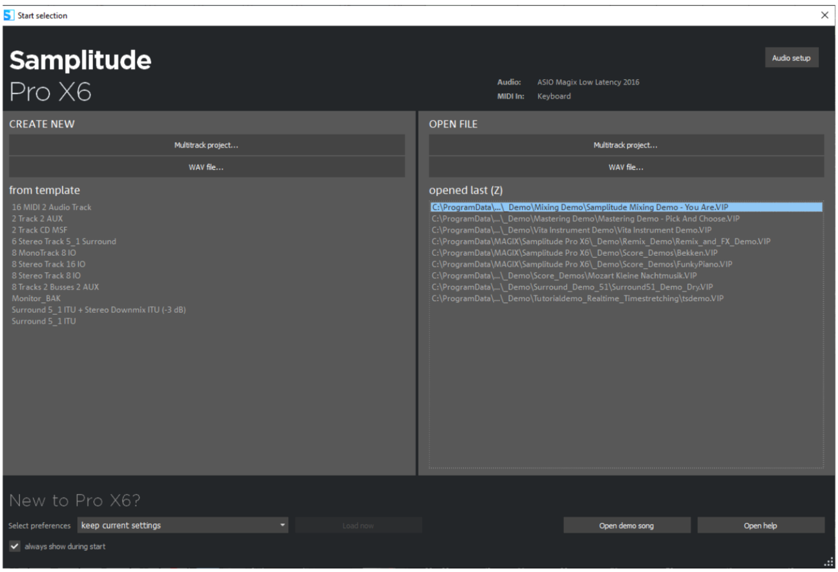
Manual Supplements

This item offers you all the latest information on current program developments.

About Samplitude

This displays copyright information and the version number of Samplitude. At the very bottom, you can see the serial number required for support purposes.

Start selection



The Start Selection opens every time you start Samplitude. You can also open it via the "Help" menu at any time. It offers several options to start working with Samplitude.

At the top the chosen **Audio and MIDI driver** is displayed, with the "**Audio Setup"** you get into the according settings dialog to change it if necessary.

On the left side are options for creating new wave projects or virtual projects. For virtual projects you have direct access to the list of project templates from the New Virtual Project (view page 563) dialog.

At the right side you can load saved projects from a file explorer window or chose from the list recently edited projects.

At the bottom of the dialog there are useful options for getting started with Samplitude with quick access to the demo project and the help file.

Load a complete set of program settings with the "**Select Preferences**" list box. More information about program settings management can be found in Options management (view page 618).

Attention: To help Samplitude beginners switching to Samplitude we changed the program behavior at some places, for instance the mouse wheel scrolls in VIPs vertically (tracks) by default instead of horizontally (time position). To make Samplitude behave exactly like in the versions before, chose "Samplitude Pro X2 Standard" from the list box. For more information, read the section Mouse Wheel at the end of Editing Keyboard Shortcuts and Menus (view page 626)

If you don't like the Start Wizard opening at each program start, just uncheck the checkbox "Always show during start".

Activating the Samplitude Dongle

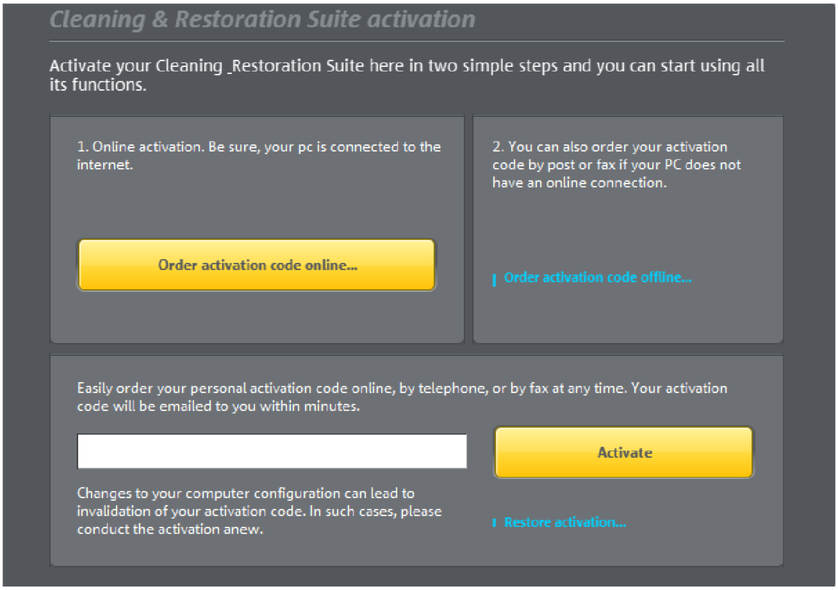
Detailed information about activation is provided in the "Installation (view page 23)" chapter.

Product Registration

Detailed information about activation is provided in the chapter "Installation > Activating Samplitude (view page 22)".

Activate Cleaning & Restoration Suite

This command activates the Cleaning & Restoration Suite.



In this dialog you can activate the Cleaning & Restoration Suite online, order an activation code offline or reactivate a previous activation.

Detailed information about activation is provided in the section "Installation > Program Activation in Samplitude (view page 22).

Download more instruments

Selecting this menu item will take you to the MAGIX website where you can browse through the latest selection of Vita Instruments and download the ones you want.

Update online

This command lets you update Samplitude at any time. This feature checks online for available updates and installs them automatically.

Language:

Here you can select the language for the program interface. Samplitude restarts in the updated language.

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