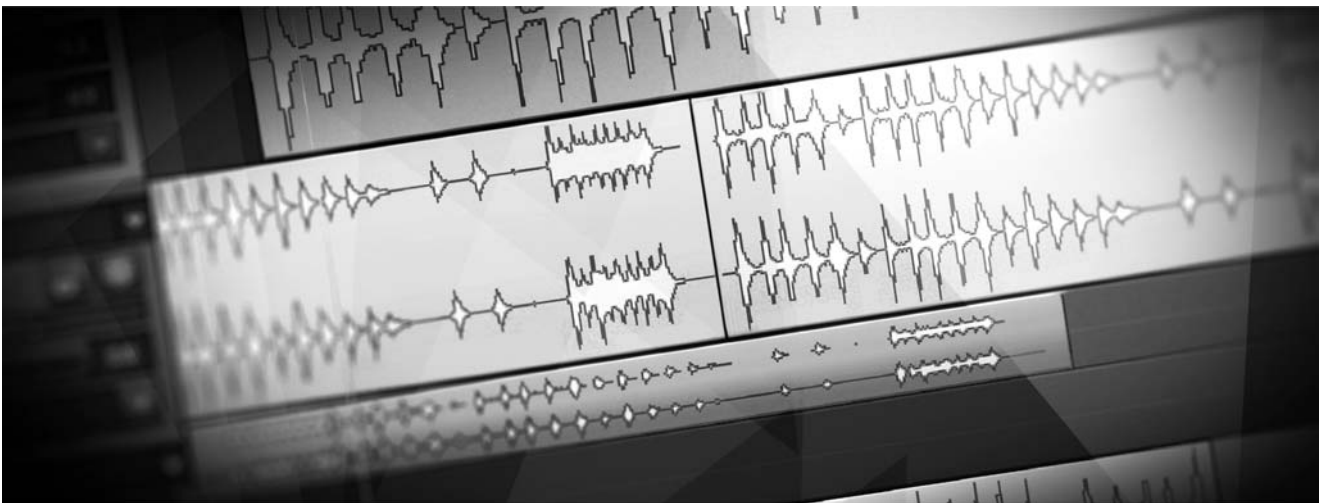




SAMPLITUDE[®]

PRO X | PRO X SUITE

the master of pro audio
recording ▶ editing ▶ mixing ▶ mastering



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This product uses MAGIX patented technology (USP 6518492) and MAGIX patent pending technology.

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Welcome

Thank you for choosing Samplitude!

You now possess 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. The program stands for unique functionality & sound neutrality, outstanding cutting & editing options, perfect CD/DVD mastering, and the flexible customization of individual workflows. Also this version was developed in close collaboration with musicians, sound engineers, producers, and users, it boasts numerous innovative functions as well as incorporating the advanced development of tried and tested performance features.

Samplitude Pro X and Samplitude Pro X Suite are now available as either a 64-bit version or 32-bit version. You can continue to rely on the superior sound and absolute phase stability of Samplitude.

In addition to this manual, we also recommend reviewing the **"Help" file**, which may be accessed via the "Help" menu; this includes more information about the individual functions in more detail. Please also use the included PDF documents (**manual.pdf**, **Effects_Plugins_and_Instruments.pdf**, **Hardware_Controller.pdf**, **Tutorials.pdf**) for further information.

You can also visit our support area and the user forum online at <http://www.samplitude.com> to find out more.

The Samplitude team

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System Requirements

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's 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 Home/Premium and 192 GB for Windows 7 Professional/Ultimate. If you work with very large projects or memory-intensive VST instruments such as samples, it is recommendable to change to a 64-bit system. 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 (soundcards, controllers etc).

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- and 64-bit plug-ins.

Be aware that certain functions and technology will only be available when using Windows x64. Find more info on our website.

Samplitude on Windows x86 (32-Bit)

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

General system requirements

- 1 GB RAM for Windows XP; 1GB RAM for Windows Vista/Windows 7 32-Bit; 4GB RAM for Windows Vista/Windows 7 64-Bit
 - Recommended: Windows 7
 - 500 MB disk space on hard disk for minimal installation
 - Graphics card with a minimum resolution: 1024 x 768
 - ASIO and/or WDM-compatible sound card
 - DVD-ROM drive
 - Optional: CD-/DVD burner, MIDI interface
- Samplitude is compatible with Windows 7.

Hard disk

The maximum number of audio tracks depends on the rotation speed, access time, and data transfer rate of the hard disk. Nowadays, modern UltraDMA hard disks are very fast and enable the simultaneous use of many audio tracks. Ideally, your hard disk should have a rotation speed of 7200 RPM (or faster) as well as an access time of under 9 milliseconds. The audio data should be saved on a hard disk separate from the operating system.

Contact

Product activation:

For questions about product activation, please contact the Samplitude service team:

Telephone: +49 (0) 5741 3455 30 (Mon - Fri 9:00 to 5:00)

Fax: +49 (0)5741 310 768

Or send an email to servicesfr@magix.net

Support

Registered users have access to technical support:

The current terms and conditions for support services apply.

You can find the terms and conditions and more information at:

<http://rdir.magix.net/?page=DNIZMJ03VM2M>

Sales

If you have questions regarding licensing and upgrades/crossgrades, please contact our sales department:

Telephone: +49 (0) 5741 3455 25

Fax: +49 (0) 5741 3107 68

Email: samplitude@magix.net

Addresses:

MAGIX AG
Friedrichstr. 200
10117 Berlin

MAGIX AG
Borsigstrasse 24
32312 Lübbecke

Installation

1. Place the installation CD into the CD-ROM drive.
2. The installation program will launch automatically. If it does not, open Explorer and click the letter of the DVD drive. Open **start.exe** by double clicking it. The contents of the installation CD are displayed, you may visit our website, or install additional programs such as CodeMeter Runtime (view page 23).



Install program:
Samplitude 11 Pro

Install accessories:
CodeMeter Runtime 32-Bit
CodeMeter Runtime 64-Bit

Also discover:
www.magix.com

Browse LAN

Welcome to the Samplitude Installer

The buttons in the navigation area to the left will allow you to select the program components you wish to install. Via the website link you will be able to access current information, i.e. the availability of new updates.

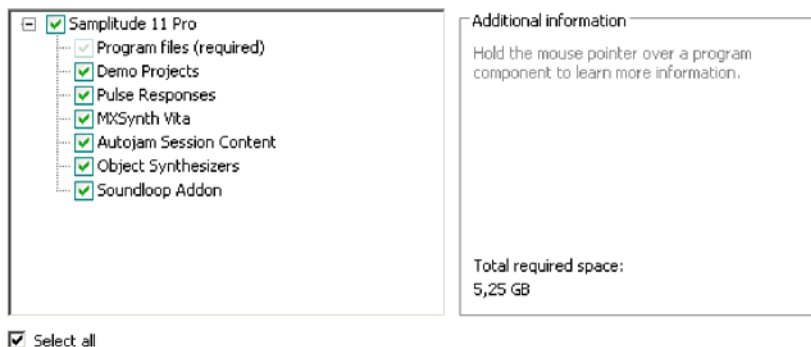
Comprehensive information regarding installation and the different versions is available in the printed or PDF handbook.

Wishing you all the best as you realise your creative ideas,

Your MAGIX Team

3. To begin the installation of Samplitude click on **"Install program > Samplitude Pro X / Samplitude Pro X Suite"**.

4. The Samplitude installation wizard will now start. Just follow the appearing dialogs. If you choose the "User defined" installation style, you can also specify the target folder for the program and choose whether you want to install additional components such as impulses responses too. In the selection screen you will see the necessary memory requirements for the installation.



After installation you can add or remove components by starting this installation program again and selecting/deselecting the respective components.

Click "**Continue**" to proceed with the installation.

Once all the files have been copied to the hard disk, a program group will be created. The installation is now complete. Complete the installation by clicking "**Finish**".

Next choose the language in which you want to install Samplitude. In the corresponding dialog you can choose between **English, German, French, Italian and Spanish**.

You are able to change language at any time after the first installation by accessing the program folder "Samplitude" and then double clicking the file "**Install.cfg**"; in the code line titled "**language=**", enter the abbreviation for the desired program language:

- **E** or **US** for English
- **D** for German
- **F** for French
- **I** for Italian
- **ES** for Spanish

If another language code is entered, Samplitude will reinstall in the language featured by your Windows system settings.

After you have saved "**Install.cfg**", the program will display the desired language version the next time it is started.

After the initial installation, you may start the program at any time via the Windows "Start" menu. Under "Programs > MAGIX > Samplitude" you can find start menu entries such as:

- Samplitude
- Samplitude Help
- README
- License terms

Activating Samplitude

After the program has started, the "Program activation" dialog appears. Select "Register".

Step 1: Enter the serial number in the next dialog. You will be able to find this on the case of the software DVD.

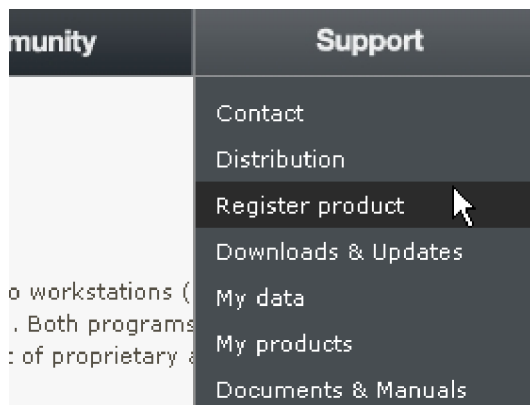
Step 2: Samplitude calculates a **user code** once the serial number has been entered.

The **actual activation code** is generated from this code later. An online connection to our authorization server is necessary for this to work. If your computer has an Internet connection, press "**Activate online**" now.

Alternatively, you can of course transfer the data onto a separate PC if your studio PC does not have an Internet connection. Simply click on the button to create a text file. You can then save it on a USB stick or any other portable medium and then copy the content (the entire URL) into the address box of a web browser.

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

You will now be directed to the **support area of the Samplitude website**. If problems occur with the automatic connection, then open your web browser, type <http://www.samplitude.com> into the address bar, and select "Register product" in the support area.



You will now proceed to online product registration. Please use your email address and password to log in. If you do not have access to the support portal, register first by filling in the required fields.

Note: Ensure sure when entering your data that your email address is correct so that we will be able to send you the activation code once the registration process has been completed.

Login

You can get support, register and activate your MAGIX Samplitude and Sequoia products here.

Login required, please login!

Email address:

Password:

Do you still need a MAGIX Samplitude/Sequoia support portal account?
[Register now!](#)

[Did you forget your password?](#)

• [Forum](#)

Language

- [Deutsch](#)
- [English](#)
- [日本語](#)

Free support for registered users:

Phone:
 +1 775 562 0528
 Mon-Fri 10am-4pm EST
 +49(0)351 4174 616
 Mon-Fri 10am-3pm CET

Enter the user code to register a new product.

Register product

Please enter your user code to register a new product. You can see your user code when the program starts or in the "Help" menu (Program Activation).

User code:
 (e.g.: 1234.5678.1234.5678)
 (or: SC11-12345-12345-12345-12345-12345):

- [Forum](#)
- [My products](#)
- [Register product](#)
- [My profile](#)
- [Download](#)

Language

- [Deutsch](#)
- [English](#)
- [日本語](#)

Click on "Continue". In the next window, continue with activation by clicking the corresponding button.

The activation code will now be displayed. This will also be sent to your email address automatically. Use the **activation code** to complete the final step:

Step 3: Now copy the **activation code** into the corresponding box in the dialog and then click on the button beside this to activate.

Samplitude is now fully authenticated and ready for unrestricted use.

On the support page under "My Products", you will now see the most recently activated product in the list of registered products:

The screenshot shows a user interface for 'My registered products'. It features a list of products with their names and version numbers, along with links for adding new products or upgrading. To the right, there is a navigation menu with options like 'Forum', 'My products', 'Register product', 'My profile', 'Download', 'Language' (with sub-options for Deutsch, English, and 日本語), and 'Logout'.

Product Name	Version	Actions
CM-Stick: [redacted]	samplitude professional 9.0 de sequoia 9.0 de	(Add new product) (Upgrade) (Upgrade)
[redacted]	Samplitude 11	(Add new product) (Upgrade)
Serial number: [redacted]	Vintage Effects Suite 1.0	(Create new activation)

- Forum
- **My products**
- Register product
- My profile
- Download

Language

- Deutsch
- English
- 日本語

- Logout

Note: Please keep your **serial number and activation code** 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 as it may change the user code. You can activate the product three times in total. After that you will have to contact our support.

Use Samplitude with CodeMeter 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 in the Menu click "Help" then "Dongle Activation".
2. An information window will inform you of the CodeMeter lease's activation. Shut the program down first.
3. Insert the dongle into a free USB port on the computer.
4. Restart the program.

Now Samplitude searches for **CodeMeter Runtime** and the license entries on startup. If your dongle is already registered, Samplitude will start and you may continue with program activation. If a license is not found, the "CodeMeter License Update" or "CodeMeter Start Wizard" will open.

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

Introduction to Samplitude network installation

Network installation – configuring the CodeMeter stick

1. Install a **current CodeMeter Runtime** on a network server or on a network PC (Windows, Linux, Mac OS X). You can find the newest Runtime at www.codemeter.de
2. Insert the CodeMeter dongle into an available **USB port** on the server.
3. Start the CodeMeter Control Center via "**Start/Programs/CodeMeter/CodeMeter Control Center**".
4. Now open "**WebAdmin**" in the CodeMeter Control Center on the bottom left. Your internet browser will open.
5. In WebAdmin, go to "**Settings**", check the option "**Start as server**", and press "**Apply**".

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

6. Return to the **CodeMeter Control Center** and exit the runtime in the menu via "**Exit file/CodeMeter**". Restart CodeMeter again by clicking "**File/Start CodeMeter**".

The server is now ready.

Network installation - Program installation

The program DVD may now be installed on a Windows client PC.

You should also install the current CodeMeter Runtime on the Client PCs in order to guarantee optimum network performance.

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

Register the dongle by first inserting it locally into a client PC or install Samplitude on the server and register straight away after starting the program. This is necessary to gain access to the latest updates on our homepage www.samplitude.com.

Please contact our MAGIX support (view page 17) if you have any questions about installation or if you require technical support.

CodeMeter Control Center

- You can find the newest Runtime in the download section at www.codemeter.de.
- Insert the CodeMeter into an available **USB port** on the server.
- Start the CodeMeter Control Center via **Start/Programs/CodeMeter/CodeMeter Control Center**.

- Now open "**WebAdmin**" in the CodeMeter Control Center on the bottom. Your internet browser will open.
- In WebAdmin go to the register **Content > CmStick**. In the **CmStick** line you will see your dongle's serial number.

- If you want to add, change and delete further licenses, click on "**Update License**" and then "Create license request". You can also create license questions or install license updates.
- Now go back to the **CodeMeter Control Center** and exit the runtime in the menu via "**Exit file/CodeMeter**".

System settings

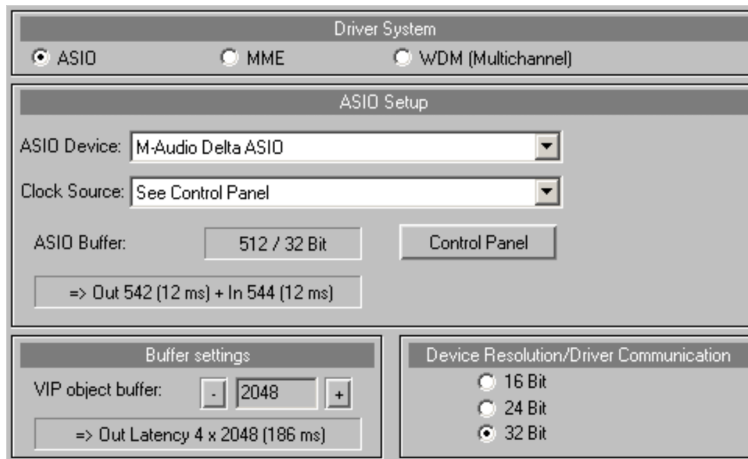
Before you start working with Samplitude, you should configure your sound card after starting the system for the first time.

Press the "Y" key on your keyboard or open the global system settings via "File > Program preferences > System/Audio".

This allows you to access information about any connected playback and recording devices as well as MIDI, metronome, or program settings, and change them if necessary. You may 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 setup

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, then you can use MME/WDM. This driver system is suitable for multi-track 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; in any case, for many sound cards, multi-track 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 provides the possibility to use the software monitoring for inputs and VST instruments.
- ASIO is intended for editing multi-track 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 sinks (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 498). For 32-bit data transfer, dithering is 24-bit or 16-bit. The settings for 32-bit are recommended especially for Samplitude

- in case no dithering is intended.
- in case the input is not fed by the sound card's converter unit, but rather directly by a DSP card.

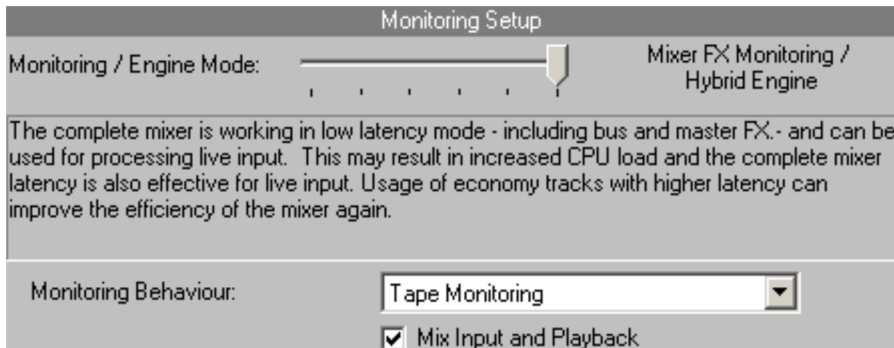
If **MME driver system** is selected, Samplitude opens the drivers with the bit depth set under **Device resolution/driver communication**. If the output device cannot process the set bit resolution, then the corresponding lower resolution will be produced and this will be transmitted to the MME driver.

To specify the **plug-in buffer** and the **path to your VST plug-ins**, switch to the dialog "System options (Y) > Effects > VST/DirectX/ReWire (view page 176)".

Monitoring settings

Monitoring is defined as listening to the input signals, for example with the intention of returning these to the incoming music. The routing of monitor signals is an important and complex process that is subject to various requirements.

In most cases, we recommend using the "Mixer FX Monitoring/Hybrid Engine".



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

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

Hybrid Audio Engine



▼ 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 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 may be played via the so-called "Economy" tracks (see below), while only the VST instrument that is being recorded is calculated by the Low Latency Engine.

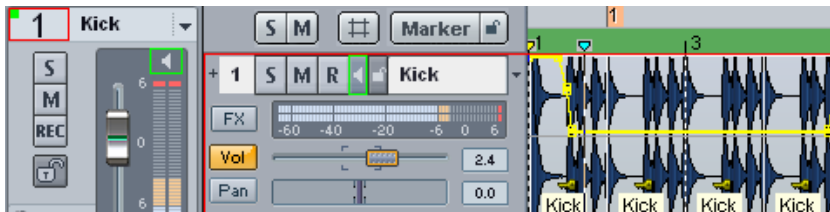
Go to **system settings (shortcut "Y")** -> "**Audio Setup**" -> "Driver System" and select "ASIO". "Monitoring settings" features the mode "Mixer FX monitoring/Hybrid Engine".

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

Economy track

If you have selected the ASIO Hybrid Engine as the driver system, you may 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 saves the system's processor from being overloaded.

To switch a track to "Economy", select the menu item "Track -> Track properties -> Economy track".



Economy tracks are indicated by a green dot in the volume display of the mixer's corresponding channel and in the track editor. The "Volume" button for the track is framed in green.

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

Overview of monitoring/engine modes

No audio monitoring (peak meter only): Displays the input level, but muted.

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

Software monitoring/economy engine: Audio monitoring includes the recording track levels and software instruments. This monitoring option is only available with ASIO drivers. No effects are applied to the input signals, and latency and CPU load remains the same, even for complex projects.

Track FX monitoring: Audio monitoring including the track effects of the recording track. This monitoring option is only available when using ASIO drivers. Connected effects may therefore be monitored in recording tracks. Bus or master effects will not be included.

Hardware monitoring/Hybrid Engine: In this case, the complete mixer operates in "Low latency" mode. This produces low playback latency. Monitoring of the input signals depends on the sound card being used.

Mixer FX monitoring/Hybrid Engine: The Hybrid Audio Engine enables audio monitoring throughout the entire mixer and also calculates playback tracks in the mixer with short

latency. You may therefore 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, since this guarantees access to all bus and master effects.

Applies to all monitoring modes: Track effects in the recording track will not be included when recording. However, the recorded track may be saved along with all track effects afterwards via the "Trackbouncing (view page 456)" function.

Note: If you use "TotalMix" from RME in combination with a hardware monitoring mode, then set the panning law in "TotalMix" to -6dB. This helps ensure that the recording level in Sampleplitude matches the monitoring levels in "TotalMix".

Mode switching

Tape monitoring (preset): In STOP and RECORD states, the input signal is played; in PLAY state, the track content is played.

Manual monitoring: With the help of the loudspeaker button in the track head or the track editor, you can manually switch on the input signal for monitoring. This mode is only effective when the ASIO driver system is used.

Mix input and playback: If you put a check mark here, then you will be able to hear the input signal during running playback too, if track monitoring is active.

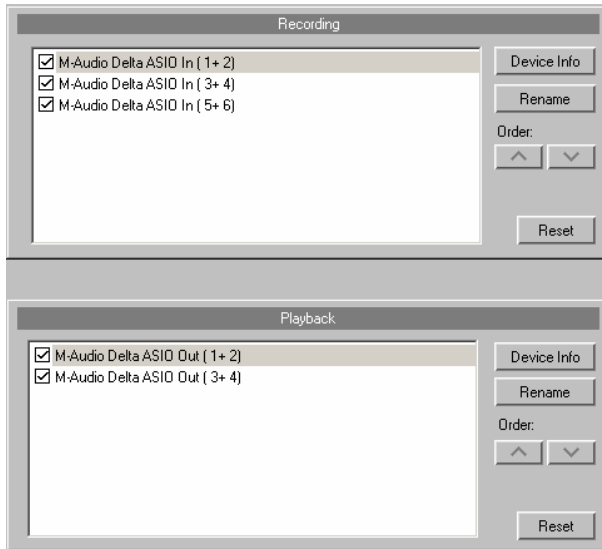
Other than the system options in the audio setup, the monitoring settings and switching processes are available to you by right clicking "**Mon**" directly. This displays two more MIDI recording options:

Automatic MIDI record switch on current track: This option ensures that MIDI tracks are always ready for recording as soon as they are selected. You can recognize this by the lit-up pink record buttons.

Automatic MIDI monitoring (thru): If this option is selected, then every MIDI track which you activate for recording will be switched on, i.e. you will always immediately hear the input signal for this track.

Audio devices

Select the inputs and outputs (devices) for the installed sound cards used by the program in this dialog.



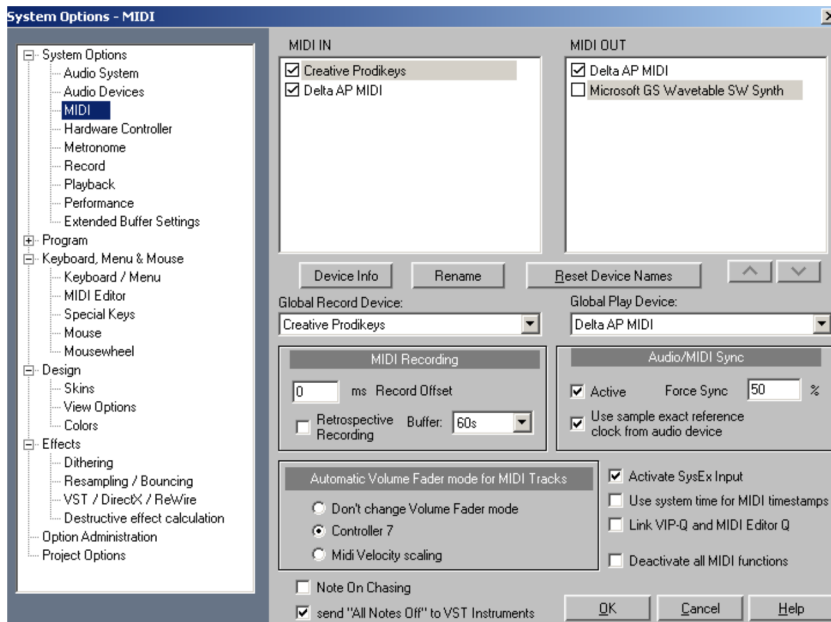
By clicking the box to the left of each respective device, you can activate and deactivate them as desired. Set the sequence with the arrow keys. The "Reset" button activates all of the devices displayed. For ASIO, only the first 4 stereo channels will be activated (standard). If you hold the "Shift" key, then all devices will be activated. Clicking the button "Device info" opens the control panel for your sound card. Press "Rename" to customize each device name individually.

MIDI setup

Samplitude offers comprehensive MIDI functions. MIDI files may be imported, inserted, edited with an external editor, and played back. In addition to MIDI timecode synchronization, the other main use is for accessing external and internal sound sources such as synthesizers and VST instruments. External hardware controllers may be accessed remotely via MIDI access signals.

Note: For seamless work with MIDI, 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** via "File > Program preferences > MIDI options" and then set the global playback and global record device. The set devices are used to play back MIDI files directly, and they are also used for 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 596)". Samplitude creates a MIDI object to the selected, recording-ready MIDI track that may 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 161).

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 then falls 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 switch off all MIDI functions in the MIDI projects or VIP projects. In this case, the MIDI menu is removed from the menu list and the MIDI functions in the arranger, track editor, and track settings dialog are no longer available.

Note On Chasing

"Note On Chasing" has the effect that held MIDI notes are also played back when they receive their "Note On" command in the arrangement before playback starts. The MIDI level activity display also for chased notes.

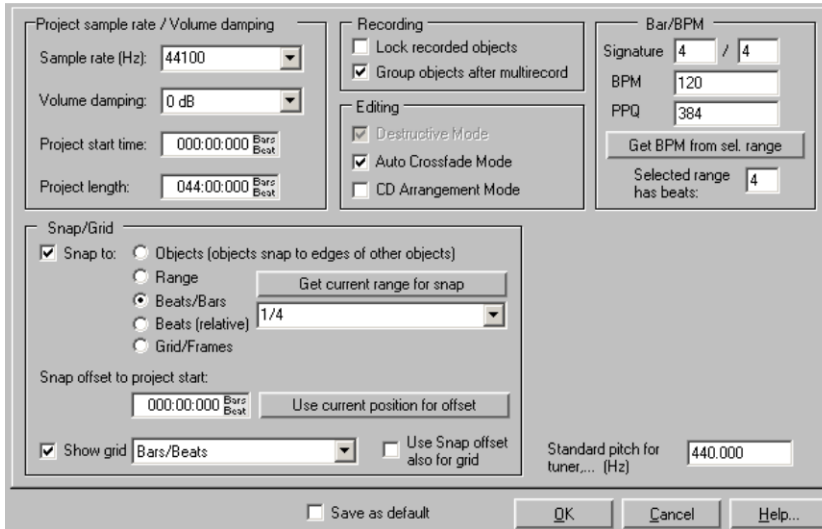
Send "All Notes Off" to VST instruments

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 systems settings.

General

Shortcut: I, Ctrl + Shift + `

Here you can see the most important information about the current project at a glance. This includes beat/BPM, recording, editing, and snap/grid settings.



Project options– Project

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.

Damping: Set level decreases in 6 dB stages. Volume reduction can be necessary when working on multitrack virtual projects if you are working with 16-bit internal precision.

0 dB editing is definitely possible when working with 32-bit float internal accuracy, since this prevents internal overmodulation. Simply set the sum levels to 0 dB via the master mixer faders.

Note: Bear in mind that Samplitude reduces the volume of wave projects by the value set in the virtual project. This way you can quickly toggle between virtual projects and wave projects without the volume level changing. If, on the other hand, a wave project is the only project opened (that is, without the virtual project it uses), it will always be played at maximum volume.

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

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

System options - Recording/Editing

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

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

- Group object according to Multirecord: Objects that belong together will be grouped in a multitrack recording.
- Offline audio editing (only for Wave projects): Detailed information about destructive editing is available in the chapter "Working in the project window -> Amplitude as a wave editor (view page 108)".
- Auto Crossfade Mode: Use this function to activate a mode which adds a crossfade to all newly recorded objects created from cuts or from wave projects which were dragged into your project. Every object may be assigned a standard fade-in, and this can be edited via the "Object editor fade" menu via "Get/set global crossfade". If two objects overlap in this mode, a realtime crossfade will occur at the intersection.
- CD arrange mode: If this is activated, Amplitude arranges newly added objects to insert a Red Book Standard-compatible pause between the objects.

Project options - Bar/BPM

Bar/BPM/PPQ: Enter the bar type (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:", then Amplitude will calculate the BPM based on the selected range when the button is pressed.

Project options – Snap/Grid

Snap to: Switches the global snap on/off.

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

Range: Activates the range grid 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.

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

Grid/frames: Activates the frame-based grid.

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 as the grid's zero position.

Show grid: If a check is placed here, then the grid will be displayed for the project according to the 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.

Project options - Standard pitch for tuner (Hz)

This field indicates a standard pitch of A at 440 Hz. You can change this value anytime you like.

What's new in Samplitude Pro X/Pro X Suite?

64 Bit Version

- Sequoia_x64.exe or Sam_x64.exe with 64 Bit memory addressing for 64 Bit-operating systems (view page 16), i. e. no 2 GB limit.
- VST plug-in handling: Automatic "bridge" handling i. e. you can load 32-bit plug-ins in the 64-bit system and vice versa.
- You can change projects from 32-bit to 64-bit.
- In the 32.bit version, by pressing the shift key, the 32-bit plug-in will run with the bridge - meaning that there is more memory available in the 32-bit version.

Restrictions:

- -Restricted video functions (only MXV and WMV)
- -No Rewire
- -VST Bridge: no VST MIDI Out Support

Tempo-Map and Tempo track

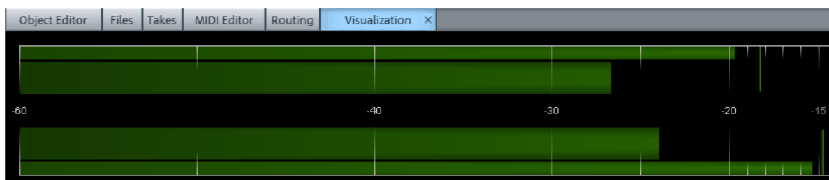
Changes in tempo can be made using the marker or graphically using a **Tempo track** (view page 305). **A tempo track contains the project's tempo map as an automation curve.**

The tempo map displays all the tempo markers and determines the musical positioning for each time position.

In the **Tempo-Map dialog** (view page 308) you can specify which objects or markers in your position should be adjusted when tempo changes are made.

Revised visualizations

The visualizations (view page 798) have been completely reworked. They have a new design.



The new interface allows you to design multi-visualizations individually. You can integrate the visualization interface into the Docking Bay (view page 51) or open it in its own window. A surround visualization (view page 806) is now also built in.

SE-cleaning effects

DeClipper SE, DeClicker/Decrackler SE, DeHisser SE, DeNoiser SE are called the new SE-Cleaning-effects. These restoration tools (view page 666) offer a range of functions that are straightforward to use, practical on a day to day basis. With as few faders and little space on the screen you get an overview of multiple instances in the project. The SE restoration tools in Samplitude don't use many resources. With the optional "Cleaning / Restoration Suite" you have access to the full versions with an enhanced range of functions.

Advanced Docking-/Manager concept

Advanced Docking-/Manager concept

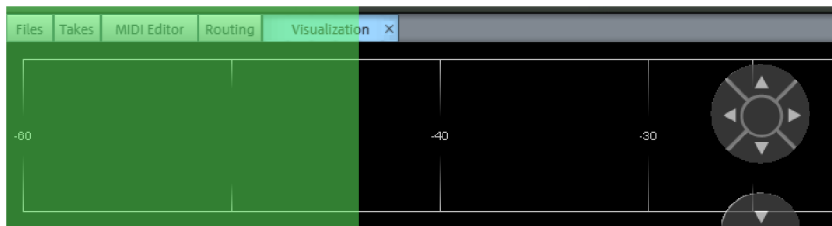
In addition to the **Manager windows** now **Visualization, Time display, Transport console, Sound pool, Object Editor** and **MIDI Editor** are grouped by default in a "Docker" (view page 50). This allows you to individually choose the display windows you need for your workflow.



Advanced Docking concept (view page 51)

Moreover you can dock certain dialogs and windows to areas in the VIP interface or to the docking bay.

Undocked windows may be docked to the areas provided in the VIP by double clicking the title bar or by clicking the title bar and moving with the "Ctrl" key pressed.



Vice versa, an individual window may be undocked by clicking the title bar or by double clicking the handle area.

Revised Object Editor

Object Editor (view page 114) and MIDI Object Editor (view page 244) have been revised and have new graphic interface designs.

- Integration in the new docker
- "To all" is now a status button:
 - Changes (Volume, Fades, EQ....) are immediately transferred
 - Changes in effect dialogs after closing the editor
 - edited plug-ins are transferred individually
 - volume changes are transferred relatively
 - EQ is always completely transferred
- Configure using right click
- Mute/Inverse button for individual channels available in the Context menu
- Volume fader up to +12 dB
- Comments field
- 4 slots for copy & paste purposes (global)
- Solo button
- Object Automation for EQ, Pan (not for 2-channel volume / pan), Object-surround (Drawing of a range is supported)
- FX presets (*.obj)
- Offline FX menu

Advanced essentialFX Suite

The essentialFX Suite (view page 685) has been endowed with 10 first-class effect plug-ins and is suited for all every day studio applications. In addition to high-quality algorithms, great importance was placed on practical icons for the individual parameters and the effectiveness of the function. The uniform operation concept, combined with low resources demands for top quality production, makes the essentialFX suite a collection of high-quality audio tools for all-around tasks.

The essentialFX Suite consists of the following plug-ins:

efx_Compressor

Classic compressor with soft clip function and advanced sidechain filter feature.

eFX_Gate

Flexible gate with optional soft knee characteristics and sidechain filter function for ducking effects.

eFX_DeEsser

Dynamic plug-in for reducing unwanted sibilants in recordings containing speech and vocals.

eFX_VocalStrip

Optimized channel strip with compressor, DeEsser, gate, Highpass filter and one shaping for recordings containing vocals or speech.

eFX_Tubestage

Authentic analog tube amplifier with oversampling mode simulation.

eFX_Reverb

Reverb plug-in for creating versatile reverb effects - from short to long reverberations.

eFX_StereoDelay

Stereo delay with individually adjustable delays for both channels, as well as tempo sync function.

eFX_ChorusFlanger

Traditional chorus/flanger effect with adjustable rate, depth, feedback and mix settings.

eFX_Phaser

Classic phaser effect with adjustable rate, depth, feedback and mix settings.

eFX_TremoloPan

Tremolo effect with optional tempo sync function and parameters for rate, shape, tremolo and panorama.

Soundpool manager

In the Docker's Soundpool Manager (view page 157) you can integrate, listen to and arrange loops effortlessly.

Independence

Included in the package is the "Independence" sampler with 12 GB of content for Samplitude Pro X and 70 GB Content for Samplitude Pro X suite or Sequoia.

This gives you access to hundreds of customized, music based software functions. The intuitive user interface and file management system, the ultra fast streaming integration, multi-core processor support and the Auto-RAM-Cleaner will enable you to store and use an huge number of instruments in seconds.

Note: Detailed information about "Independence" can be found in the PDF document.

Content Packs

- The **Content Pack 1** puts 3 GB of current drum-, percussion-, guitars-, bass-, brass-, vocals-, woodwinds-, piano-, harp- and effect-loops at your disposal for use with **Samplitude Pro X**. The loops are divided into numerous music styles and varying tempos (BPM).
- The **Content Pack 2** makes 4 GB of content categorized by music styles, instruments and BPM available for **Samplitude Pro X Suite** and **Sequoia 12**.

True Resynthesis Timestretching (TRTS) (offline)

In "True Resynthesis Time Stretching (TRTS)" the signal is broken down into individual components, with the amplitude and frequency modulated oscillators - simple synthesizers -are reproduced faithfully. If these synths play longer or shorter a Timestretching effect will be created.

You can do Timeline stretching (view page 651) using TRTS. The typical compression artifacts and associated non-uniformities will not occur. The algorithm is universally applicable. Due to linear work practices TRTS is particularly suitable for very small and very large Timestretch factors.

Spectral view

- The Spectral view (view page 496) 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 Comparison view are now available for individual VIP objects: "Object > Object color/name"
- Various colour palettes are available
- Graphical data is saved as *.hs files.

Spectral mode

Menu "Edit > Mouse mode > Spectral mode"

With the new "Spectral mode (view page 72)" you can remove noise from a track object, without the wanted signal being audibly influenced. Editing can be done directly in the Arranger window by dragging a range.

Multifunction input field with help-function

Command search bar for Finding menu commands and help topics (view page 78)

- A maximum of five results from the main menu are displayed, which can be directly selected.
- Moreover up to four results will be displayed in the help index.

Controller Artist Series support



Samplitude supports the Controller Artist Series (view page 374). Artist Mix, Artist Control and Artist Transport are all part of this. These controllers are controlled using the EUCON protocol, which can be activated in the Hardware Controller Settings (view page 360).

Note: Windows 7 is a requirement if you want to use the Artist series.

New MIDI functions

The MIDI Editor is now integrated into the Docker (view page 50) in the new version.

Transposition: (view page 253) Here you can transpose the pitch of selected events into half steps.

MIDI Timestretching: With this function you can make various changes to selected events, e.g

- double the tempo
- halve the tempo
- scale to range length.
- manual stretch. This allows you to enter individual values in the **Stretch Factor** field.

Reverse: With this function you can reverse the playback sequence. The events move by a further quantization step each time they are reversed again.

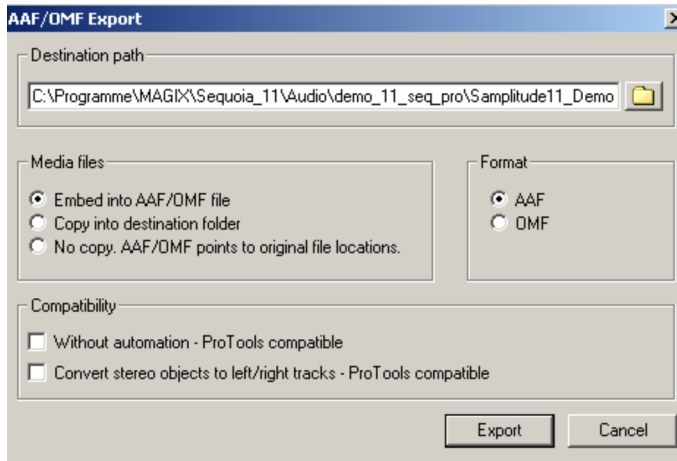
Mirror melody: This function mirrors the selected events in terms of the pitch of the re-selected event.

Further new MIDI features:

- New Snap setting **bars** (view page 66) for bar snap and quantization.
- Only one button for the mouse mode "Pen". Now there is a graphic selection menu for pen mode.
- Score anti-aliasing (binds, beams).

New functions

- Access to automation directly in the effect dialog: In many effect dialogs you now have direct access to the automation button and therefore to the automation menu. This is made possible by a newly created header (view page 179) in the respective dialogs, which also provide presets and, where needed, A / B comparison functions.
- The new command **Menu Edit > Tempo > Create click track** creates an audio track, which contains all metronome click as objects.
- BitMachine (view page 663) for creating „Lo-Fi“-Sound.
- AAF/OMF Import and Export. Export: **File > Export project as AAF/OMF (view page 461)** Import: **File > Open > Import AAF/OMF (view page 441)**



- EUCON™ Interface for Euphonix Controller Artist Series (view page 360).
- Surround functions (view page 224) for Samplitude Pro X.

What's new in 11.2?

Options administration/Remote/User administration

Note: If you work with "Remote Logins", please extract the contents from "RemoteSetupTools.zip" in your remote directory, because changes were made to "AdminCreator.exe" as well as "remote_setup_copy_from_server.bat".

- User-specific templates are now stored in "fx-preset/user_name/templates".
- If user-specific templates are available, they will only be shown to the user when creating a new project. If these templates don't exist then the templates of the local admin will be displayed.
- The auto jam session continues working with the templates in the template directory of the local admin, not with the user-specific path.
- Templates are now also saved in the INZ container. If there are no user specific templates available, the templates of the local admin will be saved.
- When importing INZ containers, previously available user-specific templates are deleted.

- "Recent Files list" and work paths for remote users are saved separately for each remote user in the "audio.ini" of the admin (in other sections)
- Current "Recent files list" is kept if you import INZ containers. You will only encounter troubles if the "audio.ini" e. g. is overwritten by an admin when loading INZ containers.
- Query when deleting a user profile in user administration.

New functions

- Cue mode: Objects that are split by crossfades of hard cuts are seen as a group.
- Clipstore: When transferring clips that contain multiple objects into the VIP, all clips will be renamed with the clip name
- Marker manager: double clicking on the marker opens the Name Editor
- Object manager: Dragging & dropping objects in the Arranger is possible
- Object manager: Context menu entry "Duplicate object"
- Root VIP editing now also for surround objects
- Hardware Controller: updated HUI templates (HUI, Mackie HUI, DM2000, O2R)

What's New in Version 11.1?

Multi-track selection

To execute the multi-track selection (view page 98) option, select multiple tracks by **clicking the respective track numbers/track ranges while holding down the "Ctrl" key or the "Shift" key.**

- **Track box context menu: "Track properties > Group track controls"**

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

In order to clear the selection, **click a track below the selection** or use the

- **Track box context menu: "Track > Ungroup track controls"**

Note: Multi-channel selection in the mixer functions exactly like multi-track 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 a channel before than the selection.

Multi-language version – one EXE for all languages

The program will be installed in the language corresponding with your Windows system settings. After installation, you may select freely between **English, German, French, Spanish, and Italian** by accessing the program folder "Samplitude" and then double clicking the file "**Install.cfg**"; in the code line titled "**language=**", enter the abbreviation for the desired program language:

- **E** or **US** for English
- **D** for German
- **F** for French
- **I** for Italian
- **ES** for Spanish

If another language code is entered, Samplitude will reinstall in the language featured by your Windows system settings.

After you have saved "**Install.cfg**", the program will display the desired language version the next time it is started.

Options management

- During loading, the current state will be saved as a "Backup" (view page 483).
- During saving, check boxes and the file name may be preset.
- Check boxes may be changed during loading.
- Loading via the "Load and restart" button.

INI patches for easily setting INI options

- INI patches: (view page 483) Collection of diverse INI entries in the "Customize" folder.
- INI patches: Load via the "Load" button in "Option administration".
- INI patches: Displays a short info text.
- INI patch: Option to apply or reset.

Metronome

- New option for metronome pre-counting with or without VIP playback: "**Play VIP during pre-count (view page 84)**"

Vandal

(Vandal in Samplitude Pro X Suite/Vandal SE in Samplitude Pro X)

- New presets and floor effects (view page 750)

Visit <http://www.vandalamps.com> for more information.

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 control 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.

MIDI

- Updated presets for "Create MIDI object": MIDI drums and MIDI phrases (filter sweep, 6 octave sequence, etc.)
- New MIDI tracks now assume the set volume fader mode (view page 161)
- Groove template selection is now in the drum editor and drum map editor as well
- <ALL> may now be selected as the MIDI input device (view page 88) for a track
- Improved SMF MIDI file import (view page 242)/export (view page 456)

System options > Design > View options > VIP

The system options also allow you to set the following parameters via "Design > View options > VIP (view page 491)":

- **CPU peak meter:** If this option is activated, the **peak meter in the track header** displays the CPU load per track. The upper peak meter displays the load on the Economy Engine for the left channel, while the peak meter below this displays the load on the Low Latency Engine for the right channel. In this case, a display value of 0dB equals a load of 100%.
- **Compressed wave display:** This display option causes the waveform to be distorted according to the zoom level. The transition from silence (less than -90 dB) to signals with lower levels (e.g. noise) is also shown more clearly.

System options -> Performance

- The maximum CPU number for audio processing may now be set via !System options > Performance (view page 477)" (new standard: 3)

New functions

- New menu command "**Tempo > Hard quantize object position**"
- New menu command "**Tempo > Soft quantize object position**"
- The "**Insert**" buttons on the **mixer (view page 158) now activate all effects** (effects inserts + plug-ins) for the individual tracks as well as in the master module. A visual indicator (*) is displayed for plug-ins that were previously active and are active the next time "Insert" button is pressed. For old projects with a deactivated "FX" button, this is activated as the individual plug-ins are deactivated.
- The upper "AUX" button in the mixer now switches all AUX paths between active and inactive, effectively making it an AUX bypass switch.
- Object mode "Lock objects (view page 77)": Holding down the "Alt" key at the same time temporarily removes the mode for the selected objects.
- CD-Text character count in the CD-Text editor
- File manager (view page 138): Folders may now be deleted in the directory tree
- Improved recording performance
- Improved synchronization (view page 347) with SMPTE and APP
- Additional improvements to details and bug fixes

Screen Elements

What is a VIP?

VIP (virtual project) refers to the basic project window for a Samplitude project. This provides a graphical framework for you to edit the project and navigate throughout the arrangement. The VIP covers all toolbars and buttons, the arranger, track header, track editor, transport console, and status display.

Furthermore, **.vip** is the file extension for virtual projects in Samplitude.

Program interface – Overview

VIP window:



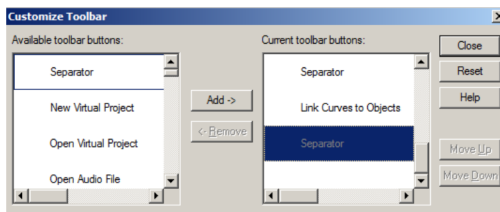
1. Title bar: The title bar is located at the top of the window. This contains the name of the application and the project. To move the entire window, simply click the title bar and move it as desired. You may also move dialog windows by dragging them by their title bar.

2. Menu bar: Menus are located directly beneath Samplitude's title bar. Every menu entry may be assigned to a keyboard shortcut. Click the icon in the above left of the mixer's title bar or in the arranger to select different skins for the program.

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 "Menu File -> Program Preferences -> Edit keyboard shortcuts and menus (view page 488)".

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 on the left separator and dragging with the mouse.

Right clicking an icon opens a **context menu**. Hide the selected bar or show large symbols in it. "**Edit Toolbar**" opens a dialog to individually remove current buttons or add other available buttons for the selected bar.



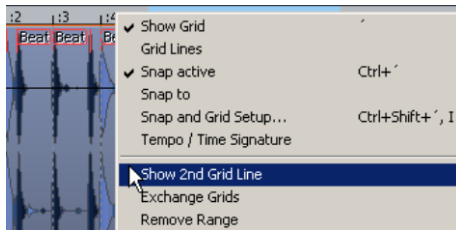
The command "**Edit Workspace**" may also be used to activate and edit the individual toolbars and to show or hide menu items.

More information about the toolbars is available in "Toolbars - Overview".

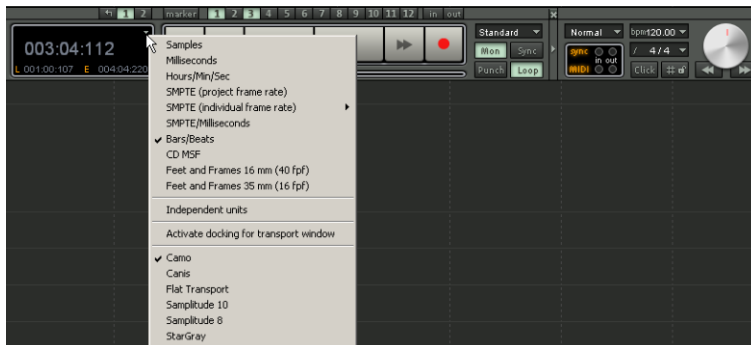
The context menu entries "**Camo**", "**Canis**", "**Star Gray**", "**Samplitude 10**", "**Samplitude 9**" and "**Samplitude 8**" provide options concerning how the buttons are displayed (view page 481).

4. Grid/marker bar: The grid/marker bars are positioned above the first track in the VIP. In the upper half (if two grid bars are displayed between both grid bars), you'll find the marker bar, where the markers and playback cursors can be positioned. The grid bars display the project time in relation to the selected measurement unit. You can also open various ranges. You can use two grid bars, for example one for beats and one for SMPTE time. To show both grids, move the play cursor by dragging with it the mouse across the upper grid. If only one grid is displayed, then the play cursor can be moved by dragging the mouse in the marker bar.

Display second grid bar: Use the grid bar's **context menu** to activate a second grid bar in the arranger and set your own measurement unit (**2. Show grid bar**).



Both grid bar positions can be exchanged (**Swap grid**). If you select "**Independent time format**" in the transport window, the upper grid will adapt to a chosen BPM grid, unlike the lower grid.



This way it's possible to select an SMPTE format independent of the project frame rate, which provides an overview of two different SMPTE displays. The option "**Activate docking for transport window**" switches on docking options for the transport window.

5. Arranger: Displays the VIP project tracks together with the audio and MIDI objects.

There are many commands for moving (scrolling (view page 101)) the visible portion of the arranger and customizing its size (zooming (view page 99)). These can be opened via the "View" menu, the grid/marker bar, and the shortcut keys.

6. Track header: The track header is at the left in front of a VIP track. Besides the track name, it contains various control elements, like mixer functions and automation. More information about the individual controls of the track header can be found in the track header (view page 60) overview.

7. Track editor

8. Setup / Zoom / Position buttons: This part of the work area helps manage each of the four different setup and zoom settings (range and zoom settings of the VIP window shown in the project clip). Similarly, the "Pos", "Len", "End", "Mouse", and "Mixer" fields can be configured by right clicking them.

9. Status bar: The status bar appears at the bottom border of the VIP window. Here, you will find about CPU overload, latency, buffer, and current operations like loading, saving, effect calculation, etc. You can open the status bar also via menu "View > Toolbars > Status bar".

10. Transport console (view page 52)

11. Manager / Docker (view page 50)

12. Visualization (view page 798)

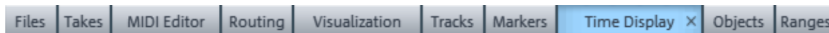
Managers / Docker

Menu "View > Managers / Docker"

In addition to the manager windows (view page 138) now visualization, time display, transport console, sound pool, object editor and MIDI editor are grouped as a tab window by default in a "Docker". This allows you to individually choose the display windows you need for your workflow.



Every docker can be minimized by minimizing an opened tab window. If you click a tab again, the docker will be maximized again.



Using the "+" key, you can open a new tab in the docker.

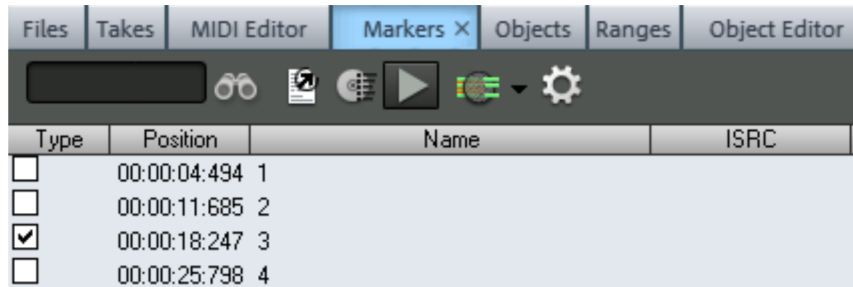


To remove a listed tab window again, just drag this out of the Docker lead or double-click on the appropriate tab window. The corresponding dialog window will now appear in its own window.

You can return the window to the docker through repeated double clicks or by dragging the window onto the tab bar.

Docking

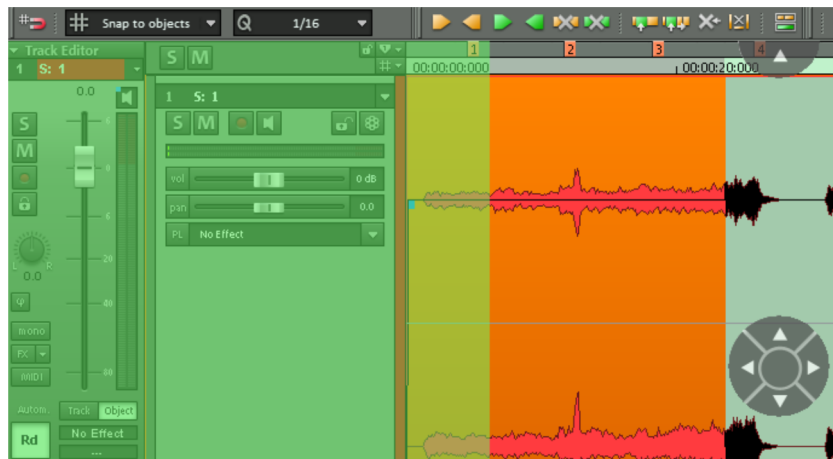
In Samplitude you can dock certain dialogs and windows to areas in the VIP interface or to the docking bay. The docking handle ranges for docking and undocking are found at the top of the respective window and, in a docked state are distinguished from the selected skin by a bar or double line.



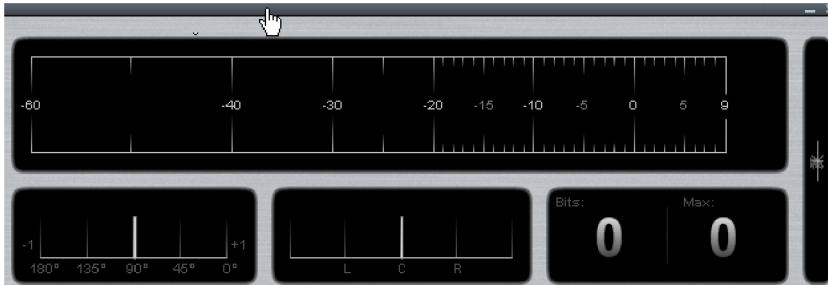
In a undocked state the corresponding dialog appears with a conventional title bar.



Undocked windows may be docked to the areas provided in the VIP by double clicking the title bar or by clicking the title bar and moving with the "Ctrl" key pressed. When moving windows an arrow symbol appears on the interface. Move an undocked window with the mouse pointer onto one of the arrow symbols this results in the area which docks when you release being displayed as a transparent rectangle.



Vice versa, an individual window may be uncoupled by clicking the title bar or by double clicking the handle area.



Also inside the docking bay you can place and arrange new windows. Move the respective window using the mouse pointer onto one of the arrow symbols in the docking bay.



The docking function is available for the following windows:

- Manager
- Visualization
- Time display
- Soundpool
- Object Editor
- MIDI-Editor
- Transport console

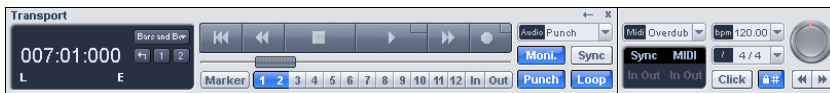
Transport console

Shortcut: Ctrl + Shift + T



Transport console with "Camo" skin

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



Transport console with "Star Gray" skin

Play / Stop / Fast forward / Rewind buttons: Use these buttons to control the positioning like a cassette player.

Right mouse click on play button: Opens the playback parameter window for stereo master operation. This allows you to specify the sample rate, playback device, and scrubbing/varipitch options, etc.

Right mouse click on record button: This opens a dialog for setting recording options. You may also start stereo or mono recording from here directly.

Record button: This button begins recording for all active tracks, i. e. tracks featuring a red "R" button in the track box.

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

Range buttons: You can save ranges with buttons "1" and "2". Using the arrow symbol you can open previously used ranges.

L/E time display: Displays the length and end position of an area and can be edited by double clicking.

"Marker" button: This button opens the marker manager for further marker editing.

More information can be found in the chapter "Manager -> Marker manager" (view page 147).

Marker buttons 1...12: The current play position may be saved to one of the 12 marker buttons by clicking. If one position has been saved, the marker will appear bright. Another click on the same marker moves the play position to the corresponding marker. Right-clicking deletes the saved marker again, making it available again to be allocated.

Recording modes: Here you can set up the appropriate recording mode.

Standard mode (playback while recording): This is the typical recording mode for multi-track productions. Here, the currently active tracks are recorded. The other tracks are also played back.

Record without playback: If this option is selected, playback of existing objects is deactivated during recording, and the play cursor remains at the beginning of the recording. Playback may be started manually, for example in case "read after write" is required. Set the play cursor at a different position and start playback by pressing the "Play" button. The recording will not be interrupted by doing so. This way, changes to previously recorded material can be made without affecting the recording process.

Only Sequoia: In record mode **Record without playback/read after write** You can use the export function even during recording.

Punch marker mode: In this mode, only the range between the punch start marker and punch end marker is recorded. This setting corresponds to the "Punch" button (see below).

Auto Jam Session: Opens the "Auto Jam Session" window.

Detailed information about "Auto JamSession" is available in the reference manual, the help file, and in the "manual.pdf" documentation under "Playback -> Auto JamSession (view page 591)"

"Mon" button: This button activates the Samplitude's record monitoring feature; all tracks with an active "R" button display the relevant input signals in the peak meters. Right clicking the "REC M" button lets you select from the various monitoring modes.

Please also read the chapter "System options > Global audio options > Monitoring settings (view page 27)" for more details.

"Sync" button: This button opens the dialog with the synchronization setup.

More information is available in the chapter "Synchronization (view page 345)".

"Punch" button: This button switches Samplitude into Punch Marker mode. In this mode, only the range between the punch starter marker and the punch end marker is recorded.

If you have not previously created a punch marker, one will be set automatically when you press the record button. If the recording is started when punch markers already exist, the

recording button will flash until the start marker has been reached as well as once the end marker has been passed. It will light continuously during the actual recording between the markers. End the punch recording by pressing the record button again; playback will not be interrupted. Playback will stop if you press the "Stop" button.

"In" button: This button sets the starting point for a Punch recording.

"Out" button: This button sets the end point for a Punch recording.

Additional Punch In/Out markers can be set by holding down the **Alt key**.

"Loop" button: Use this to switch into Loop mode, that is, a specific range is played repeatedly.

MIDI record modes: The following MIDI record modes are available: normal, overdub, multi-overdub, and replace. The different modes determine how the newly recorded MIDI files will be added to the VIP.

More information can be found in the chapter "Samplitudequick start -> Workshop: recording (view page 80)".

Tempo section: The tempo section displays the tempo at the current position. You may change the playback speed and the tempo of the complete arrangement. All objects in the VIP are adjusted to the most recent speed value by moving them or stretching them.



The "Beat grid/snap (view page 66)" button enables you to activate and display the beat grid.



Use the "CLICK" button to switch on the metronome click. Right clicking the "CLICK" button opens the metronome options window (view page 84) to configure the pre-counter and click volume.

Scrub control: The scrub control wheel can be used to adjust playback speed. This can be used to find audio passages.

The buttons beneath it can be used to start playback forwards and backwards at a slower speed. This too can be used to improve control over audio passages, like editing out crackles or other errors later.

Track Editor

Located at the left border of the arrangement window, the track editor enables quick access to all of the most important track parameters of 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 via this view. All relevant settings of the corresponding track may be viewed without opening the mixer or the track view in the arranger window.

Open the track editor via the corresponding track button at the bottom-left of the arranger window beside the "Workspace" selection field or via the menu item "View -> Track editor".



- 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. "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 171)".

3. **"M" button:** This button mutes the active track (track mute function).
4. **The "REC" button:** This activates the track for recording.
5. **The "Lock" button:** Protects objects in the track and prevents them from being unintentionally moved or deleted.
6. **The "Panorama" dial:** This controls the position within the mix. Right click one of these two control elements and the stereo panorama dialog (view page 220) will open to adjust additional settings like panning laws or the stereo width.
7. **The "Phase reverse" button** reverses the signal phase 180 degrees. Right click on one of these two control elements and the stereo panorama dialog (view page 220) will open allowing you to adjust additional settings like panning laws or the stereo width. Detailed information about the stereo editor can be found in the chapter "Stereo panorama dialog (view page 220)".
8. **The "Mono" button:** This switches the track to mono processing from the input to the pan controller. In particular, all track effects preceding the pan controller operate in mono, which saves considerable CPU resources. The submix and AUX return buses always remain stereo, however. If stereo objects are located in mono tracks, the mono share (L+R) is generally played.
 Right click one of these two control elements and the stereo panorama dialog (view page 220) will open to adjust additional settings like panning laws or the stereo width. 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, may be freely adjusted in the "FX routing" dialog. If AUX sends are used, the AUX send panorama controller can be used for panning the mono signals.
9. **The "FX" button:** Opens the dialog for specifying the effect sequences (view page 174) and adding VST/DirectX plug-ins; right clicking this button also provides access to the track effects settings for the active track. These may be copied, inserted, reset, saved, or loaded. Save your personal track effect settings in the program directory in "FX presets -> Track FX". Of course, new subfolders may also be created. We've already included a selection of useful presets, e. g. "Mid-side processing". The track effects settings of a VST instrument may also be saved (including parameters and all subsequent track effects), and transferred to other tracks.
10. **The "MIDI" button** This switches the track to MIDI recording and opens the MIDI section (view page 58) of the track editor.
11. **Volume input box and volume controller.**
12. **The loudspeaker symbol** switches on monitoring/playback of the incoming signals when the "REC" button is active and if "Manual monitoring" is selected in the system options. "MIDI thru" will be active here for MIDI tracks.
13. **Control display:** Shows both of the track's LED chains for the input and output signal.
14. **The "Automation" button:** This activates automation (view page 327) for the track.
15. **Automations parameter selection field and automations control:** Select automation parameters and adjust the values with the corresponding controllers.



1. **Arrow:** Opens and closes the dialog box.
2. **In:** Opens the MIDI Input device menu.
3. **Out:** Opens the MIDI Output device menu.
4. **Channel in:** Set the MIDI Input channel.
5. **Channel out:** Set the MIDI Output channel.
6. **Program:** This button is used in MIDI mode for program selection of 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:** Set the MSB byte for the "Bank select" MIDI message for controlling your external instruments here.
8. **Bank LSB:** Set the LSB byte for the "Bank select" MIDI message for controlling your external instruments here. The bank number is MSB value* 128 + LSB value.
9. **Drum map:** Here you can select a drum map (view page 270) 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 of the respective MIDI input or output up or down here. This function works in real time, meaning that the MIDI files in the MIDI objects will not be changed.
11. **Velocity dyn:** This button activates velocity dynamics as a real-time 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 255)".

12. **Input Q:** In this case, the MIDI quantize settings (view page 259) are used to quantize every recording offline. The original position may be restored via the menu item "Object -> Quantize -> Reset Quantization".

Audio



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

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.

5. Plug-ins: You can use FX inserts, MAGIX plug-ins, DirectX, and VST effects for this track. Click the button to the right of the field beside "Plug-ins" to activate and open the "DirectX/VST plug-ins (view page 174)" dialog to specify a plug-in setup for this track.

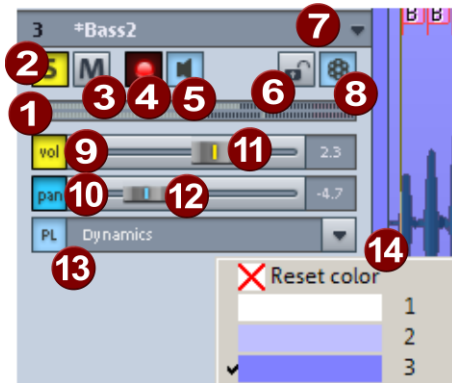
6. AUX: Specify the input for the aux sends (view page 172) here

7. EQ: Contains the parametric EQ for this track. Right clicking opens a convenient input window.

8. Comments: Track info section for quickly adding notes on tracks.

Track header

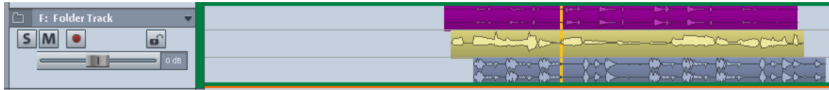
1. **Control display:** Shows both LED chains for the input and output signal for the track.
2. **The "S" button** (view page 171): Mutes all tracks except the selected one (track solo function).
3. **"M" button:** This button mutes the active track.
4. **The "R" button** (view page 82): This activates the track for recording.
5. **The loudspeaker symbol** switches on monitoring (view page 27).
6. **The "Lock" button:** This protects objects in the track and prevents them from being unintentionally moved or deleted.



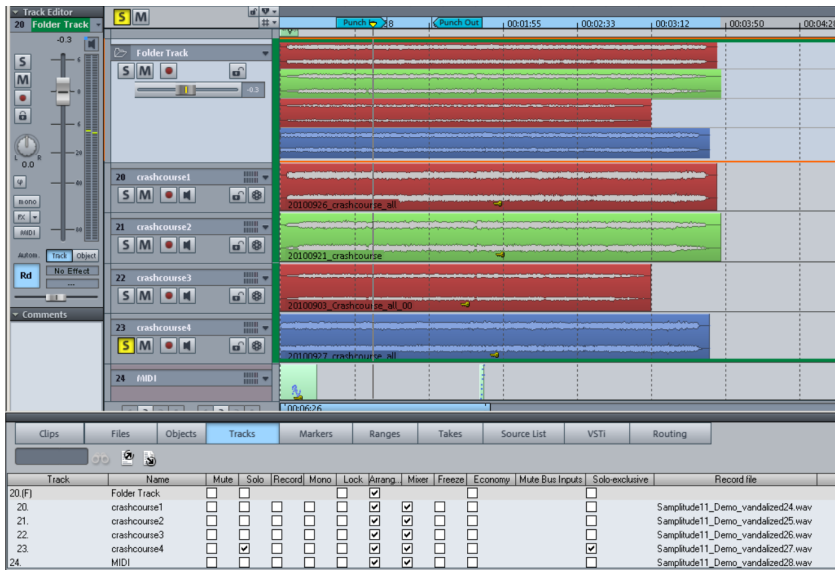
7. Clicking the track name or the track number selects the track. Repeated clicking the track number causes the track's display height to switch between flat and high. Double clicking on the track names enables them to be edited. If you want to add or remove a track from the multi-track selection (view page 98), then hold down "Ctrl" while clicking the track name. Right click the track name to open the **track settings** dialog; click the arrow to select additional track-relevant functions from the context menu.
8. **Revolver tracks:** Revolver tracks (view page 62) may be used to compile objects differently for each individual track. The combinations created for the corresponding track may be accessed at any time in the revolver tracks menu.
9. **Vol:** This button switches on the level automation curve (view page 327). This allows level adjustments in the track to be controlled via an automation curve and drawn by moving the channel fader.
10. **Pan:** Switch on panorama automation with this button.
11. **Volume** controller
12. **Panorama** controller (view page 220)
13. **PL:** The plug-in selector (view page 92) assigns various **insert effects, VST, or DirectX plug-ins** to a track.
14. **Color selection** (view page 74): If the right edge of a track header is clicked, a selection menu will appear for you to specify the color of the track and its objects.

Folder tracks

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.

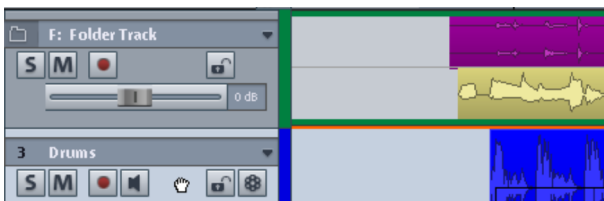


The tracks in the folder track are displayed within a frame, and the folder track number is appended with an "F" in the track manager.



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.

Tracks can be moved into the folder via drag & drop by clicking on a free area in the corresponding track box. The mouse pointer turns into a hand. Now drag 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 copy folder tracks.

The following functions work for all tracks found in the folder track:

- Hide them via the track minimization button (-/+ symbols in the track header of the track folder). All tracks included in the track folder are displayed in reduced size.
- Mute, lock, and solo
- Volume regulation All individual volume fader tracks are scaled relative to each other.
- Focus for recording

All tracks in the folder track are displayed when the folder track is open. Each track receives 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.

Section marking in the folder track is carried out across all tracks. You can use this function for section-based cut operations across all tracks of 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 of this track are displayed in the folder and may be used for simultaneous object-based cutting operations.

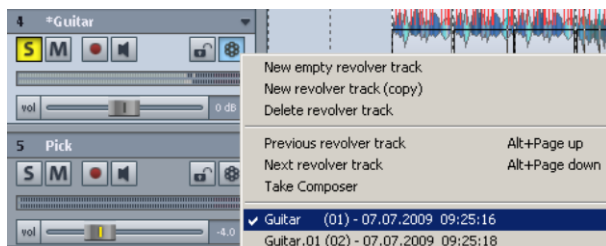
Working with folder tracks and reference tracks:

1. Initiate a multi-track recording. This creates several grouped objects, one below the other.
2. Select an area over all the objects that you wish to edit together. You can now create a folder track.
3. Minimize the folder track and choose one of the contained tracks as a reference track.
4. The object of the reference track can be seen now in the folder track instead of the emblematic display of all contained tracks. All object editings, cuts, fades, and object editor effects for this object are now applied to all objects of the track folder.

Reverber tracks

In Samplitude Revolver tracks are at your disposal. Revolver tracks can be used to compile objects differently for each individual track. The combinations created for the corresponding track may 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. If the new **"Camo"** or **"Canis"** skin is being used, then a button specifically for this function is present beside the lock symbol.



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

Of course, the context menu may also be used to create a new, empty revolver track, to delete the current one, or to display the the previous or next revolver track (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.

Revolver Tracks are stored in the "**TrackData**" project subfolder.

To edit or view existing revolver tracks in the overview, open the take composer (view page 152) 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.

Workspaces

The purpose of workspaces is to sort menu entries and toolbars in such a manner that you have a good overview of the functions of Samplitude. Workspaces bundle commands for certain tasks like mastering, editing, or recording.

As well as the **Hide menu entries** option (File > Program Preferences > Edit keyboard shortcuts and menu... > Hide menu) and **redesigning toolbars** (right mouse click on the toolbar), you can also save your **settings as a presets**.

You will see the selection box for the workspace at the bottom left corner of the Arranger window. Some workspaces are already predefined. "Power User" shows collectively both toolbar lists and commands.

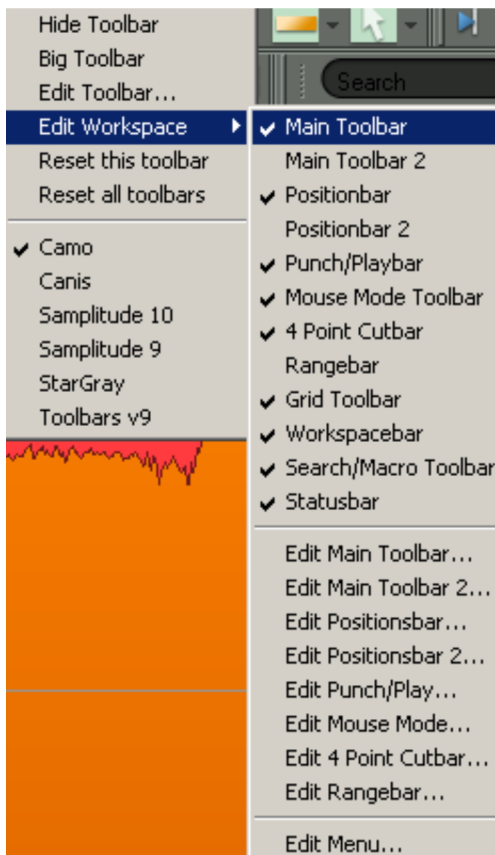
Creating a new workspace: To create a new workspace, open the context menu by **right clicking on the workspace bar** and then selecting "**New workspace**". You will now be asked to enter a name for your workspace. The new workspace contains all settings of the previously activated workspace as well as your current changes. All further adjustments are automatically saved in the workspace. Manual saving is not required

Adjust the workspace: Open the context menu and click "**Edit Workspace**". Here you can select which toolbar you want to have displayed in your new workspace. **You can activate or deactivate each bar individually, or add/remove individual symbols.** You may also **hide menu items** with the "Edit menu" command. Simply select the corresponding menu item from the keyboard shortcuts and then press the button "**Show menu item**" or "**Hide menu item**".

Toolbars – Overview

All toolbars may be moved freely on the screen. To do so, click on the separator at the left edge of the group of icons and drag the toolbar to the desired location. These will snap in at different positions in the window. For example, you can also easily swap the position bar with the toolbar.

Right clicking on an icon enables the skin for the buttons to be swapped, the size of the buttons to be increased, shows or hides tools, or otherwise customizes the different toolbars. You can also access the dialog "Edit Keyboard Shortcuts, Menu, and Mouse" via "File > Program Preferences > "Edit Keyboard Shortcuts and Menu". The menus may be individually edited via the corresponding buttons by either showing or hiding the different items.



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 "Menu File -> Program Preferences -> Edit Keyboard Shortcuts and Menu (view page 488)".

To quickly **sort a toolbar's individual buttons**, click on the corresponding icon and drag it to the desired position while **holding down the "Alt" key**. You can use the same method to remove icons from the toolbar by holding the "Alt" key, dragging the icon outside the boundaries of the toolbar, and then releasing it.

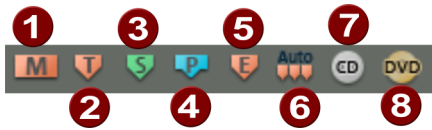
The following overview shows the presets:

Toolbar: (left section)



- 1 New Virtual Project
- 2 Load virtual project
- 3 Load audio file
- 4 saving
- 5 Cut
- 6 Copy
- 7 Insert clip
- 8 Split Object
- 9 Glue objects
- 10 Undo the last action
- 11 Undo the last action
- 12 Auto crossfade mode
- 13 Crossfade Editor
- 14 grouping
- 15 Cancel grouping
- 16 Mixer

Toolbar: (right section)



- 1 Set Marker
- 2 Set CD track index
- 3 Set CD sub index
- 4 Set CD pause index
- 5 Set CD end index
- 6 Automatically set CD track Markers
- 7 Create CD
- 8 Create DVD Audio

Grid toolbar/snap buttons



The grid toolbar is available in the toolbar. This enables important changes to be made to the snap settings, without always needing to access the project options dialog (view page 33). Click the magnet button to activate the snap grid. Activating the button to the right opens a context menu with the following snap settings options available:

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

Snap to range: Activates the range grid 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.

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

Grid/frames: Activates the frame-based grid.

Specify the snap length (for beat snapping only). Values range from **every 4 beats, every 2 beats, every beat, musical measure** and **same as quantization**.

The setting **Musical measure** 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.

Note: When snap is activated the play cursor jumps in the transport console according to the snap values, even when using the fast forward and rewind buttons. If you would like to temporarily shut off the transport controls, press the "Alt" key simultaneously.

Snap active: Switches the global snap on/off.

Shortcut: Ctrl + #

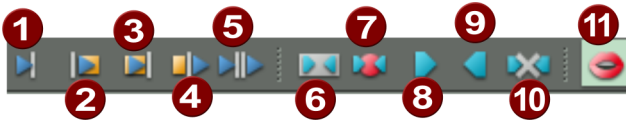
Snap and grid setup: This option jumps to the **Project options - general** (view page 33) dialog to specify settings for the snap and grid.

The area indicated by a **Q** to the side you can specify the quantization value used for MIDI (view page 259) and audio quantization (view page 550), "**same as quantization**" sets the snap setting accordingly. This causes audio and MIDI quantization to be linked (view page 33). Right clicking the field opens the MIDI quantize settings.

Note about "Input Q": In the **interface skins** (e.g. "**Camo**"), you will see the button "**Input Q**" in the MIDI area of the track editor (beside the button for velocity dynamics). If this function is active, then recorded MIDI notes will be quantized according to the current settings immediately. The original position may be restored at any time via the menu item "Object -> Quantize -> Reset Quantization".

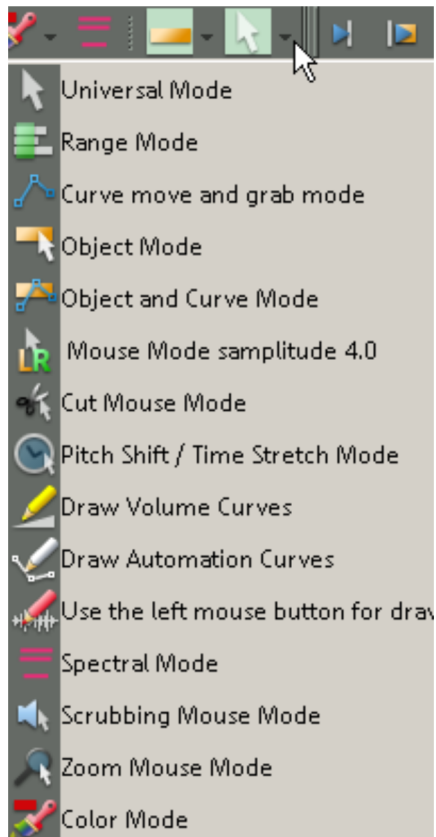
Details about audio quantization are provided in the menu reference "Object > Quantize > Audio quantization wizard (view page 550)".

Punch / Playback bar



- 1 Play to cut start
- 2 Play from cut start
- 3 Play to cut end
- 4 Play from cut end
- 5 Play beyond cut
- 6 Punch In mode
- 7 Punch recording
- 8 Set punch in marker
- 9 Set punch out marker
- 10 Delete punch marker
- 11 Cue mode: Playback stops after each object

Mouse mode bar



Universal Mode

This is the preset mouse mode in Samplitude. All necessary functions are available by left clicking. Depending on the relative position within the track, various functions will be executed. Right clicking always opens a context menu.

The vertical position of the mouse distinguishes between object handling and range manipulation within every VIP track: In the upper half, you can select ranges and set the play cursor position. In the lower half, objects may be selected and moved.

Range Mode

In this mode, only ranges and the play cursor (position line) may be manipulated.

Additional functions are also available via the following keys:

Period key "." - Press this button to switch Samplitude to "Object" mode temporarily. Objects may now be selected and moved.

Hyphen key "-" - Press this button to switch Samplitude to "Curve" mode temporarily. The volume and panorama curves may now be edited.

These special functions enable all important functions to be reached quickly, and accidental movement of an object cannot occur.

Curve Mode

This mode is strictly for editing automation curves (view page 327). This mode may be opened temporarily via "Universal/range" mode by pressing and holding down the hyphen "-" key.

Place the cursor over a curve and click with the left mouse button to set a new curve point. Curve points that have already been set can be moved easily using drag & drop.

In case you want to 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.

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 may 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.

Detailed information about how to assign keyboard shortcuts for temporary function 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 488)".

Object/Curve Mode

In this mode, objects may be moved and curves may be edited with the left mouse button.

Mouse Mode Samplitude 4.0

Use this function to switch to the Samplitude 4.0 mouse mode. The right mouse button then controls object functions, and the left one controls range manipulations.

Detailed information about how the functions of the individual mouse modes is provided in the menu reference, help file, or in the "manual.pdf" documentation under "Menu > File options -> Program settings -> VIP mouse mode" (view page 505).

Cut Mode

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

Under "System options -> Keyboard/menu -> Special keys -> Temporary key for cut mode (view page 488)", you can specify a keyboard shortcut to keep "Object cut" mode active (provided the shortcut is held down).

Pitchshift/timestretch mode

The object handles below to the right and those in the object centre can be used for directly setting the playback speed and pitch. Advanced functions are available for Tempo marker manipulation in the timeline (view page 511).

Detailed information about how to use "Timestretch/pitchshift" mouse mode can be found in the menu reference, help file, or documentation "manual.pdf" under "Edit > Tempo > Tempo Map Dialog (view page 308)"

Volume Draw Mode

This mode allows volume automation curves (view page 327) to be drawn with the left mouse button.

Automation Draw Mode

In order to draw an automation curve or a MIDI controller curve, select "Automation Draw Mode". Selected curve points can be deleted using the "Del" key. If you move individual points using the "Alt" key, the bordering can be lifted using the neighboring points to the right. Holding down the "Shift" key while drawing fader movements produces a soft curve.

Object automation

Select "Object automation" in the track editor and place a check mark for the desired parameters. The object curves for the object automation (view page 333) will be added. Now draw the corresponding automation curve by dragging with the mouse.

Track automation

This mode allows automation curves to be drawn for the selected track (view page 328) by using the left mouse button. Activate the automated parameters of the embedded plug-in beforehand with the shortcut "**Ctrl + Alt**" **move the corresponding plug-in's handle** as desired. Next, click the "Track automation" field in the track editor and place a check mark for the desired parameters in the context menu. Now you can draw the curve of the selected parameter onto the track ("Automation Draw Mode").

The selected parameter's curve can be seen in the track in the foreground; all other curves may be displayed as thin lines by selecting this option in the context menu.

This mode also enables panorama curves, etc. to be drawn with the left mouse button. The panorama curve on the respective track can be deactivated and activated again anytime with the "Pan" button to the left in the track head.

Tip: You may also move any plug-in parameter at any time via the small horizontal faders in the track editor's automation box while in an automation drawing mode mode (e.g. touch, latch, overwrite, or trim).

Draw waveform in the wave window

In "Wave editing" mode, use the left mouse button to draw 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 remove very short impulse disturbances manually.

Spectral mode

Menu "Edit > Mouse mode > Spectral mode"

With the new "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.

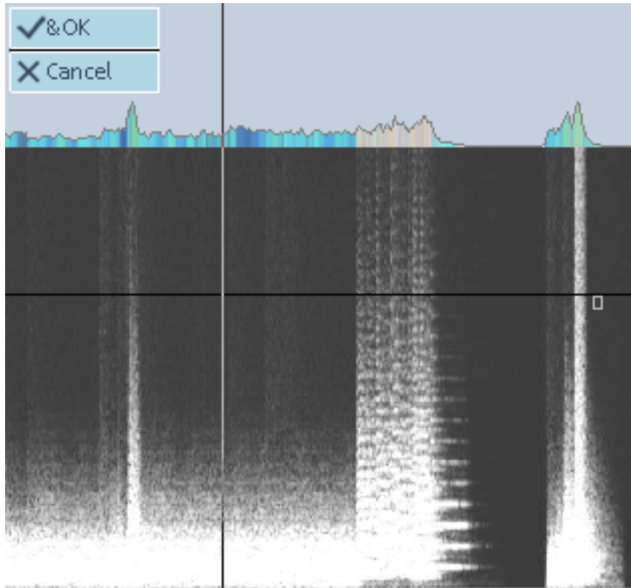
To be able to work with "Spectral mode" particularly effectively set the Spectral display (view page 496) in display options.

The graphical interpretation of the music is displayed in the Spectral display using 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.

After calling up "Spectral mode" highlight the unwanted noise by drawing a rectangle around it using the mouse. With help from the handle you can adjust the selected area.



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. After making the correction, you can see the results in the Wave/Spectral display in the Arranger window.

Scrub Mouse Mode

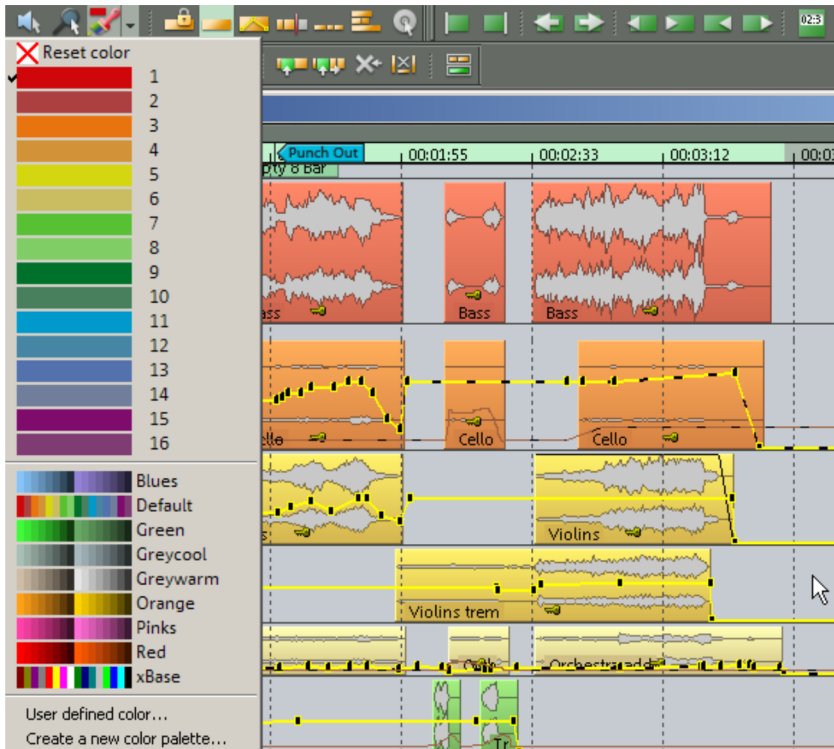
This mouse mode enables you to monitor and control the playback speed (view page 107). In this case, the project may be played back in the forwards or reverse scrubbing direction.

Zoom Mode

use the right mouse button to zoom out of the project, and click with the left button to zoom into it (view page 99).

Color mode

Color mode enables you to 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 to a paint bucket symbol. 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 not color the object, but rather the **wave form** instead.

If you hold down the **"Ctrl"** key while doing this, then you will not color the object itself, but rather the **object's background** instead.

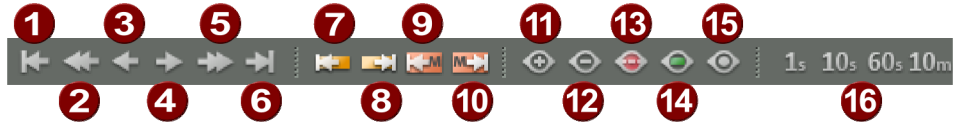
If the preset color selection is not sufficient, then you may also **select your own color** or even **create your own personal color palette**.

The option is also available to access the color palette for each track by clicking the color selector on the right edge of the track header; this also enables a new color to be assigned to a track. If multiple tracks are selected (view page 98) 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 object's wave form color

Position bar 1



- 1 Section to start
- 2 One section left
- 3 Half section left
- 4 Half section right
- 5 One section right
- 6 Section to end
- 7 Play cursor to previous object edge
- 8 Play cursor to next object edge
- 9 Play cursor to previous marker
- 10 Play cursor to next marker
- 11 Increase clip size (zoom in)
- 12 Reduce clip size (zoom out)
- 13 Show all
- 14 Zoom to range
- 15 1 pixel = 1 sample
- 16 Predefined zoom range:
1 second, 10 seconds, 60 seconds, 1 minute

Position bar 2



- 1 Freely defined zoom areas 1-4
- 2 Increase section vertically (zoom in)
- 3 Reduce section vertically (zoom out)
- 4 Show all vertically
- 5 Show range vertically
- 6 Zoom into waveform
- 7 Zoom out of waveform
- 8 Overview mode

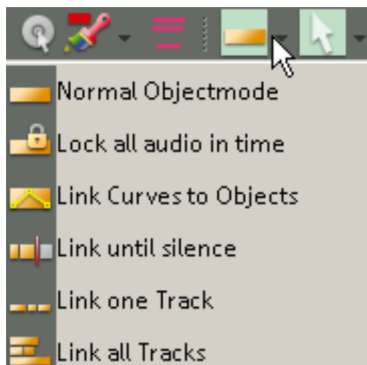
Range bar



- 1 Play cursor back to range start
- 2 Play cursor to range end
- 3 Fold range to left
- 4 Flip Range right
- 5 Range start to previous zero crossover
- 6 Range start to next zero crossover
- 7 Range end to previous zero stage
- 8 Range end to next zero stage

- 9 Range Editor
- 10 Move object start left
- 11 Move object start right
- 12 Move object(s) left
- 13 Move object(s) right
- 14 Move object end left
- 15 Move object end right
- 16 Step size for moving objects

Object modes:



Normal Object Mode

This mode moves objects and edits the end and start positions, fade in and out phases, and object volume.

"Lock all audio in time"

This mode locks all objects to stop them from being moved accidentally. Object mode "Lock all audio in time": Holding down the "Alt" key at the same time temporarily removes the mode for the selected objects.

Link curves with objects

In this mode you can shift track automation curves when moving objects together with the objects.

Link objects until pause

The means that the chosen object is selected and moved together with all subsequent objects on this track. This way partial ranges of a project, if separated by pauses, remain unaffected from moves.

This function also applies to multiple selected objects arranged along various tracks.

Link objects on active track

In this mode, all objects on the current track are selected and moved together.

Link objects on all tracks

In this mode, all objects appearing in the arrangement are selected and moved together from the mouse position.

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

More information about using the object modes is available in the chapter "Working with object modes (view page 123)".

Multifunction input field with help-function

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



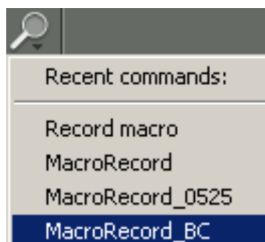
Enter a character combination into the search field which is relevant to a search term. Immediately up to five commands from the main menu will be displayed that contain the desired character combination.

Moreover, in the section below you will see up to four help topics that contain your search terms.



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

By clicking on the appropriate help topic the program help will open up displaying explanations from selected topic areas. You can see a list of recent commands by clicking on the little arrow of the magnifying glass.



You can actually execute a command by clicking on the commands that are listed.

Samplitude Quick Start

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 play cursor, click on the desired position in the timeline.
- To move the playback cursor, use the left and right arrow keys.
- In order to jump to the most recent position, press the back space key
- To move the VIP under the play cursor, use the keyboard shortcut Alt + Arrow left / Alt + Arrow right.
- To create a range, use the mouse to draw in the 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 cursor, select the shortcut keys Ctrl + Q / Ctrl + W.
- To create markers at the play cursor position, select "Shift + 1...0" in the letter keys block on your keyboard. 0" im Buchstabenblock Ihrer Tastatur
- To jump to a marker position with the play cursor, select the corresponding number 1...0 in the letter key block of your keyboard. 0" im Buchstabenblock Ihrer Tastatur

Detailed explanations of numerous additional navigation functions can be found in the chapters: "Working in the project window", "Object-oriented audio editing (view page 123)" and in the menu reference, help file, or complete documentation "manual.pdf" under "Edit > Range (view page 519)". A complete overview of all keyboard shortcuts can be found in the chapter "Preset keyboard shortcuts".

Workshop: Recording

This chapter features information about recording audio material in Samplitude. This explains the basic functions involved.

Creating a VIP

First of all, create a new project by clicking the "New multi-track project (VIP)" button in the start wizard on program launch. If you already have Samplitude open, select "New Virtual Project (VIP)..." in the "File" menu.

The screenshot shows the 'New Project' dialog box with the following settings:

- Name: NEW
- File Path: C:\...I\X\Samplitude_10_Pro\Audio\NEW
- Create New Project Subdirectory
- Presets:
 - Project Template: -----
 - Mixer Setup: [0] Stereo Master
 - Surround Setup: (empty)
- Project Options ... (button)
- Track Number:
 - 1 Track
 - 2 Tracks
 - 4 Tracks
 - 8 Tracks
 - Custom: 16
- Sample Rate: 44100 Hz
- Default Project Length:
 - 1 min
 - 60 min
 - 10 min
 - 74 min

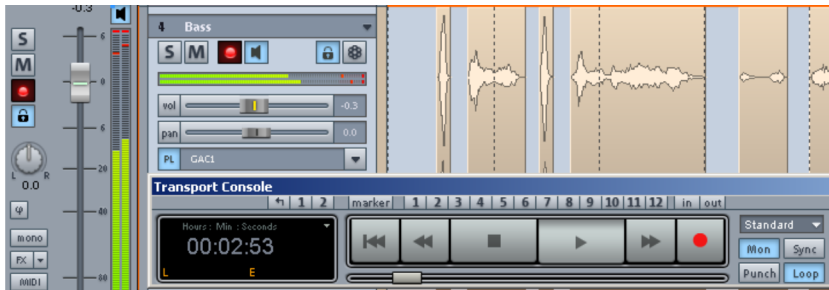
Name your new project and select the file path where you would like to save the VIP. You may also create a new folder where all files relating to this project may be saved. Select "[0] Stereo Master" as the **mixer setup**.

In the "Track number" field, enter how many tracks you wish to use. This doesn't mean that the project is limited to this number; new tracks may be added at any time. Adapt the "Sample rate" to the sample rate supported by your sound card and confirm it with "OK".

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

Preparing to record

To record from your internal CD/DVD drive or from an external sound source using a cable/microphone connection, it's important that your **sound card is connected to the sound source**. You can check this by switching on the track which you wish 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.



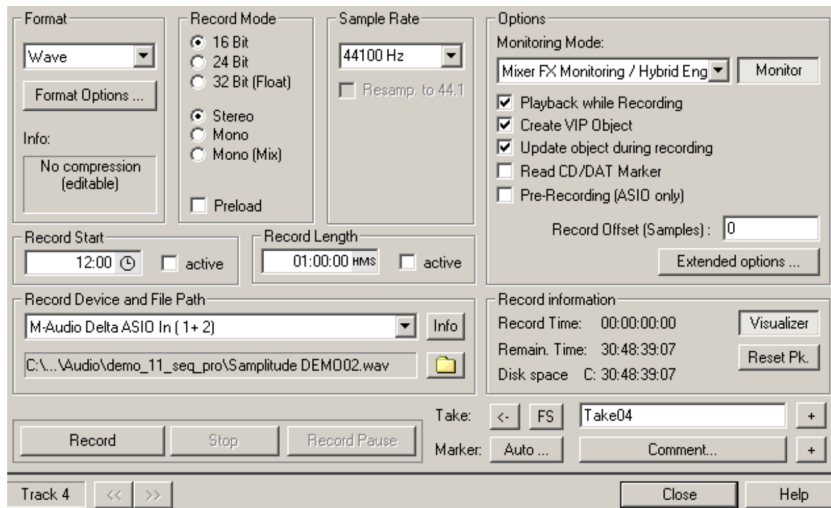
If you can't see the input level peak meter, check in the track editor (menu "View -> Track editor") to ensure the **sound device is connected to the same port listed under "Audio -> In"**. If this is not the case, set the track's input device to the port of the sound card input of the sound source by clicking "Audio -> In".

If the level is in fact visible as a bar but sound is not yet audible, then make sure that the **output device of the track correlates to the output of your sound card**.

If you have selected "Manual Monitoring" as your monitoring mode (Shortcut "Y -> Audio Setup"), please **click on the loudspeaker symbol next to the level display on the track in order to view and hear the input signal**.

Now right click the record button in the transport window.

This will open up **Record Options**:



Select the **format** for the recording first. Detailed settings for the desired audio format available via "Format options".

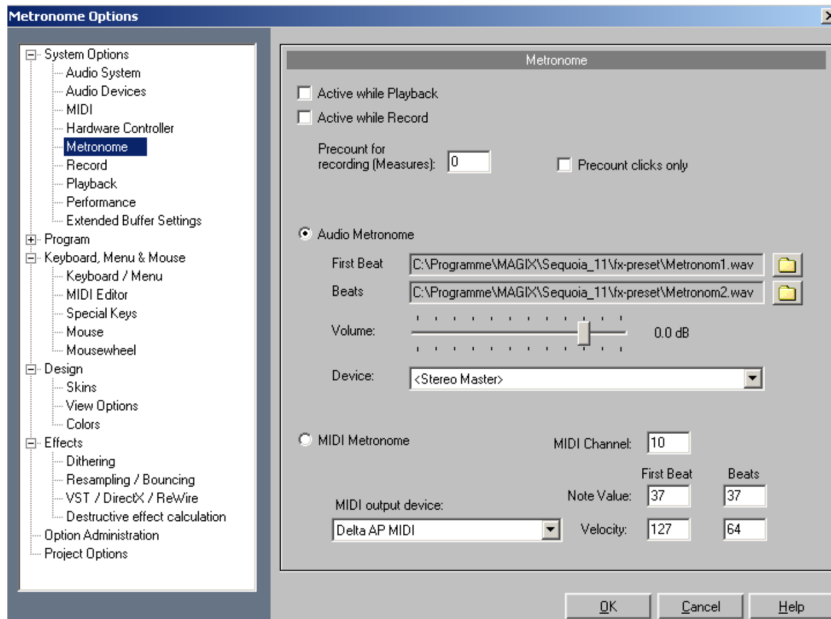
If **record mode** is set to 16-bit at a **sample rate** of 44.1 kHz, then this is equivalent to CD quality. A bit resolution of 24-bit or even 32-bit (float) makes sure that hissing during subsequent editing of the sound material remains at a minimum. On the other hand, the higher the bit resolution, the more space that is required.

Now specify if the **track should be recorded in mono or stereo**. While a stereo recording mode is usually selected for CD/DVD recordings, consider whether it would be better for single instruments, like bass or guitar, to be recorded as mono tracks, since this could improve future sound editing and addition of the instrument into the mix. 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 -> Record options (view page 582)"

Metronome settings

The dialog "Options > Program settings > Metronome settings" allows you to activate the metronome function. The metronome may also be set as an audio metronome or as a MIDI metronome.



Active while playback: This option activates the metronome click during playback.

Active while record: This activates the metronome click during recording.

Pre-count for recording (measures): Specify the number of beats that the metronome will pre-count prior to recording. If you activate "**Pre-count clicks only**", the metronome will stop before recording.

The "Punch recording" button in the toolbar (view page 67)" blinks during pre-counting and the "Record" button blinks in the transport console (view page 52). When recording begins, both buttons remain active until the end of the respective recording period.

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 581)".

Audio metronome

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

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

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

Device: Set the audio device for the metronome here. **Stereo master** is preset, but this may be changed to another audio device output as the source of the metronome.

MIDI metronome

MIDI output device: Set the device that should produce the metronome click here. This is normally the sound card.

MIDI channel: Set the MIDI channel via which the MIDI commands are sent here.

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.

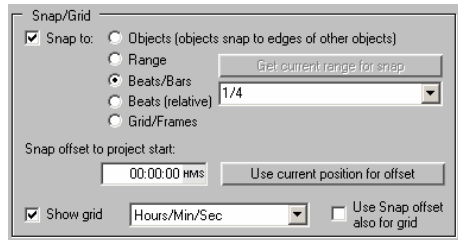
Volume Adjustment

Click the "**Visualizer**" button under the recording information display. Various functions are presented to aid visual orientation. Select "**Setup > Peak Meter**". If there is no visual display of levels is shown, then press the "Monitor" button under "Options" in the recording dialog. You should now be able to see a signal.

Detailed information about "Visualization" is available in the reference manual, the help file, and in the "manual.pdf" documentation under "View > Visualization (view page 798)"

Note: Connect your input signal to your external amplifier/your sound card's input in such a way that the peak meter is as close as possible to 0db, but doesn't quite reach this value.

The **recording starts from the current position of the play cursor**. If you wish to record from the fifth bar (for example), place the play cursor at the beginning of the fifth bar. To do so, open the **project options under "View > Grid > Snap and grid setup" (shortcut: "I")** and check the "Snap to", "Grid/Bars" and "Show grid" boxes. In the "Show grid" "selection menu, select "Bars/beats" as the grid unit, and confirm 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. Switch the **display** in the **transport window** (view page 52) to "**Beats**". 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. Simultaneously, you can read the bar and beat position in the play cursor of the transport window.

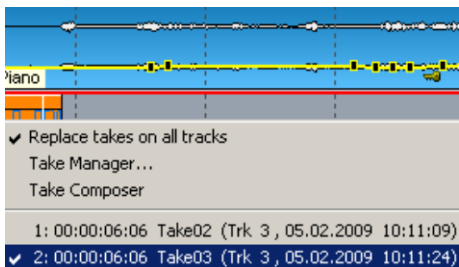
If you click on the **"Record" button now; recording will start**. As soon as you have recorded enough, press the stop button. You will be asked if you wish to save your recording. If you are happy with this, press the "Yes" button.

Your recording appears in the project window as a virtual object. Now activate the next track by clicking on the **"R" 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 in case settings have already been specified and record straight into the VIP. Simply press the "R" key on the keyboard. If a track is activated, recording will begin immediately.

Loop recordings

If you would like to **record in a loop**, select the section in the arranger that you want to record first. Activate the "Loop" button in the transport control and start the recording. The range will repeat until recording is ended. A new take will be created for every run. If you hold down "Ctrl" and click with the right mouse button on the last take created for the recorded track, you will be able to see all of the takes produced using loop recording. To listen to a specific take, select it from the menu that is displayed.



The Take manager (view page 150) is useful for organizing and editing recording runs. Use it together with the Take composer (view page 152) to produce the perfect take.

Punch recording with markers

If you know the exact range in which a punch recording prior to recording, then use punch recording with markers. To do this, **select the area with the mouse where you want to use punch recording**, and then activate the "In" and "Out" buttons in the **transport control** above the red "Record" button. Position the play cursor next.

If "Punch" is activated, **start the process with the "Record" button on the transport console**. The actual recording will occur within the punch range. **While this is happening, the "Record" button will flash when the play cursor is in front of the "In" marker. The button remains red during the punch recording.**

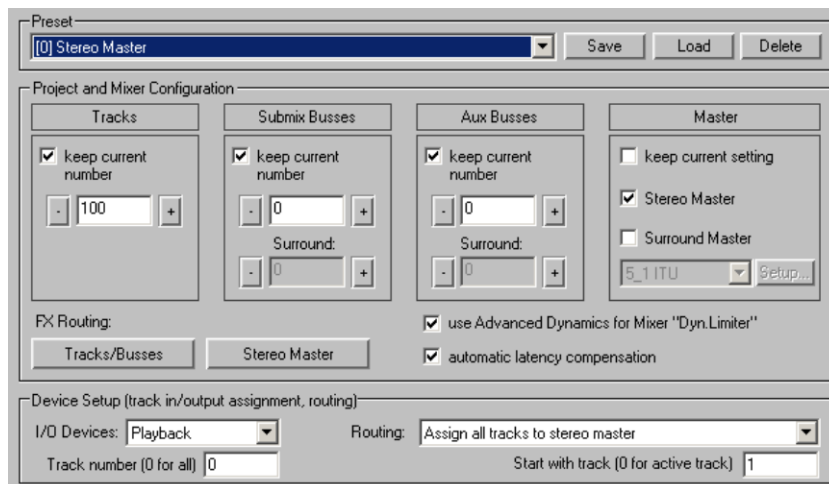
Example: An error between beats 23 and 25 needs to be corrected. Playback starts before the punch in position so that the introduction to the recording is made as easy as possible. During the punch process the "Record" button flashes; the recording will start automatically from the "In" marker (bar 23). Since the section following the object is already satisfactory, the recording is ended automatically at the "Out" marker (bar 25).

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 581)".

Multi-track recording

The "**Project options - mixer setup**" dialog can be very useful for configuring multiple tracks in multi-track recordings in a single step.

1. Open "Project options - mixer setup" via the key combination "**Ctrl + Shift + M**". You can also access it via "Options -> Project properties -> Mixer setup...".
2. Click the "Routing" box and select the option "**Route all tracks to existing stereo devices**" if you want to record in stereo. For **mono recordings**, select the option "**Route all tracks directly to existing mono devices**".
3. If the recording doesn't start at the current track, enter the corresponding track number into the "**Start with track**" field.
4. In the "Device setup (track in/output assignment, routing)" area, click the "**I/O devices**" box and select "**Record**" from the dropdown list.
5. For example, if you have four recording devices, enter "4" into the "**Track number**" field. The devices will then be set up across the next four tracks.
6. Click "OK" to confirm.



Preparing to record

- For each track you want to record on, press the "R" button to activate the track, i.e. to prepare it for recording.
- Click on the button with the loudspeaker to activate monitoring.
- You can set the maximum gain of your signal from your recording sources. To monitor the peak levels, open the peak meter via each track's context menu (track header -> "Track visualization").

Starting recording

Once you have everything set up and the signal has been adjusted, **start recording by pressing "R"**. **End recording by pressing the space bar** (only in "Playback while recording" mode) – **or by pressing the "R" key again**. Accept the recording by pressing the "OK" button in the dialog window that opens.

Now you will see one object for each recorded track.

MIDI recording

In Samplitude, there is **basically no separation between audio and MIDI tracks**. Each track may contain audio and MIDI objects. You may therefore work on audio and MIDI material in a single project without having to split tracks. This enables you to manage VST instruments entirely from within a single track. When freezing the track, the MIDI data is converted into audio data. However, **a track may only record from a certain device**, which means that audio and MIDI files cannot be recorded simultaneously onto one track.

In a Samplitude VIP track, MIDI data is recorded in the same way as audio data. An object is created for each recording, which overlaps existing objects.

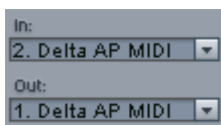
Note: Samplitude offers special MIDI recording modes (view page 90), in which already existing MIDI objects are inserted either via mixing (overdub) or via local replacement (replace).

Preparing a MIDI recording

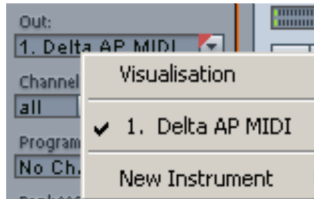
If the "MIDI" button is activated in the track editor, the MIDI section opens to prepare the track for MIDI recording.

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

If you have integrated multiple MIDI input devices with your system, you may also select <ALL> as the MIDI input device for a track.



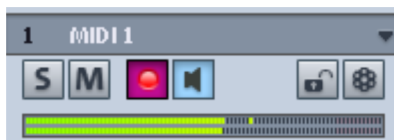
As the playback device (MIDI output device), select the MIDI port you use via the **MIDI out slot** (e.g. MIDI out of your sound card or a VST instrument).



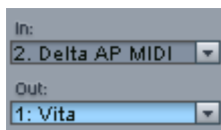
If sound cannot be heard when the keys on your device are played, then monitoring likely needs to be activated. Activate this via the loudspeaker button (MIDI thru) for the respective MIDI track.



By right clicking the **"Mon"** button in the transport control provides access to the behavior of the **"Record"** and **"Monitoring"** button. If you activate the options **"Automatic MIDI record switch on current track"** and **"Automatic MIDI monitoring (Thru)"**, then the respectively 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, then 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.



Ensure that your MIDI synthesizer is sending to the same MIDI channel that you have selected in the "channel out" slot in the track editor.

Note: Many drum computers 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 recording modes

The following MIDI record modes are available: **normal**, **overdub**, and **replace**. The different modes determine how the newly recorded MIDI files will be added to the VIP.

You can specify the settings for MIDI Record Modes in the transport control. Click on the arrow on the left beside the "Close" button in the transport control to display this section.

Alternatively, you can select the MIDI recording modes from "MIDI record" mode via "Play / Rec > MIDI record mode".

Normal: Recording mode matches that of the audio recording. This means that a new MIDI object is created over the existing object for each recording. The old object remains, but it may 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 150) later on.

Overdub: The data is recorded into an already existing object, available and newly recorded MIDI data is mixed together.

Replace: The data is recorded to an existing object, and any existing MIDI data 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.

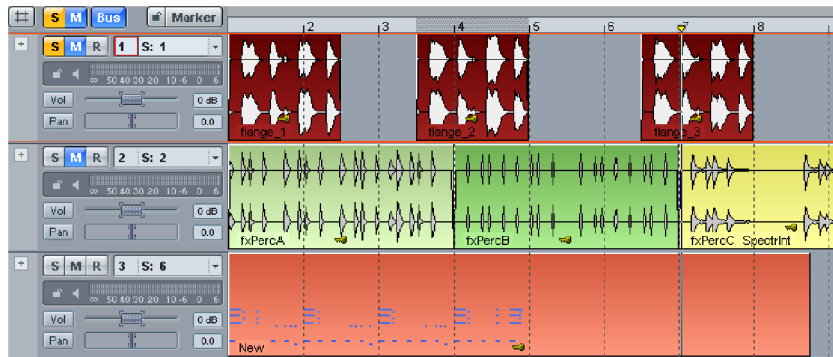


Now you can now start the MIDI recording using the shortcut key "R". After ending recording with the "Stop" button, you will be asked whether you want to keep or delete the recording. After confirming this with "OK", your newly recorded material will appear as a "MIDI take" in the VIP. In case you have 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 150).

Note: You can switch between the individual recording passes even quicker by holding the "Ctrl" button down and then right clicking 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), then you can follow the recording via 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.

Objects in the virtual project (VIP)



Audio and MIDI

Audio objects in a VIP always reference one wave file or one partial section of a wave file. MIDI objects also reference a single corresponding MIDI file. In contrast to audio, in which case wave files are saved visibly on the hard disk, MIDI files are saved together with their objects, which links them directly to these objects.

Cutting and editing

Audio and MIDI objects are cut in the same way.

Place the play cursor at the cut position. Now click the object to be cut and press the "T" key on the keyboard. If "Auto crossfade" mode is activated, then Samplitude will create a fade for audio objects between the two newly created objects.

Detailed information on this can be found in the chapters: "Object-oriented audio editing (view page 113)", "Object editor (view page 114)", "Working with objects (view page 123)" and "MIDI in Samplitude (view page 242)".

Mixer



Use the mixer (keyboard shortcut "M") to adjust the volume and panorama of the recorded tracks as well as automate fader and panorama movements. There are also multiple plugins, 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, not only for each individual channel, but also for the complete stereo track.

Detailed information is available in the chapter "**Mixer** (view page 158)".

Effects

Note: You can find more detailed information on effects in the accompanying PDF ("Manual" and "Effects, plug-ins and instruments").

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

Offline effects

The effects can be used in Wave projects and on objects. A Wave project represents an audio file. Objects in a virtual project refer to this audio file. Offline effects change the audio data in the wave project and may be set via "Effects -> Process Effects Offline".

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, without changing your original audio material.

What effects are there?

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

- **Dynamics**
- **Frequency / Filter**
- **Delay / Reverb**
- **Distortion**
- **Restoration**
- **Stereo / Phase**
- **Modulation / Special**
- **MAGIX Plug-ins**
- **essentialFX**
- **DirectX** (only available if DirectX effects are installed)
- **VST FX** (only available should the VST effects be installed)
- **ReWire** (only available for tracks and if ReWire clients are installed)
- **Analog Modeling Suite:** am-track, am-pulse, am-phia (Samplitude Pro X Suite and Sequoia only)
- **am-munition** (Samplitude Pro X Suite and Sequoia only)
- **Vintage Effects Suite:** CORVEX (Chorus, Flanger), ECOX (Echo, Delay), FILTOX (Modulation, Filter). These are special internal plug-in effects for dynamics, sound editing, and simulation of analog switching.

Effects - Signal flow

Offline effects are applied before all real-time effects, as 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 real-time effects in the following order:

1. Object effects
2. Track effects
3. Master effects

Effects in audio objects

Object-based virtual effects are set in the object editor. These may be opened by double clicking on an audio object. The effect view is opened here by default. To display the full-size object editor, place a check next to "Max".

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-related effects may be set either in the VIP window or in the mixer. These may be internal effects or VST FX or DirectX effects.

Now click on the arrow symbol for the corresponding field. A menu with the various effect categories will open.



Mixer effects

To set track effects in the mixer, open the mixer by pressing the "M" key. In the "Ins" section, click the arrow symbol of an effect slot in the corresponding channel. Selecting effects is done the same way as described above. Use the "FX" button directly below the track's channel fader to open a dialog to change the series of effects within the track.



Master effects

Master effects are applied to the entire material in a multi-track project. The signal is routed through the effect, i.e. your audio card, before reaching the master output.

Master effects may only be set in the mixer, which may be opened by pressing "M". The sum range of the mixer is wider than normal tracks and can be identified via the differently colored faders.

The effect slots for MAGIX plug-ins as well as VST and DirectX effects are located at the very top. All of Samplitude's internal master effects are located under this. The equalizer may be used directly from the mixer or the equalizer dialog, which may also be opened by right clicking one of the EQ controls.

The **sequence of the master effects** may be set in the **FX routing dialog**, which opens via the FX button beneath the master faders.

CD mastering

CDs may 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.

Set CD tracks

Position the play cursor where you wish to set the CD track marker. The marker menu is located above the track head boxes in the VIP. Open this and select "Set CD track index". Repeat this process until all markers that you want to set are in place.

Now position the play cursor where you want the CD to end. Open the marker menu again and select "Set CD end marker".

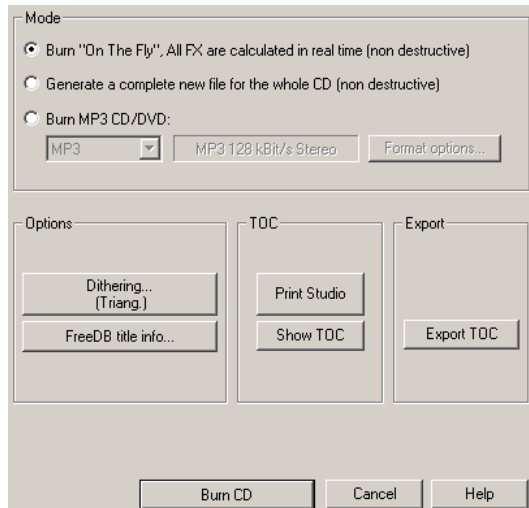
Note: The burning process only starts from the first CD track marker.

Burn CD

Now play the project again and at Samplitude's 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 may 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 777)

General Functions in the Project Window

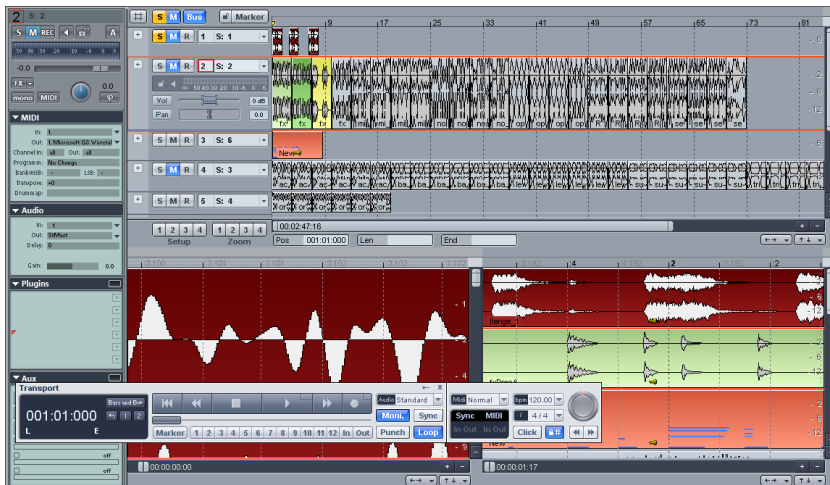
Section

"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.

Up to three different sections may be displayed at a time in a project (shortcut: "B"). This allows you to display the entire project in the upper section while special passages of your arrangement are displayed in the other sections, e.g. at loop start and loop end.

The simplified view may be restored via the keyboard shortcut "Shift + B".



Active section

If you display your project in several parts only one can be shown at a time. Activate a part by clicking on it or clicking on its control. By clicking on both double arrow buttons at the left, lower border of every section a corresponding context menu will appear that offers you different possibilities for moving the corresponding section horizontally or vertically.



The red arrows in the lower toolbar are additional buttons for moving sections horizontally.



Detailed information about sections is available in "View -> Sections (view page 811)" and "Edit > Range > Split range (view page 522)".

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, the selected automation mode, 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 multi-track selection function is also available in the mixer for previously added control groups (view page 167) if the same control elements are integrated.

In order to clear the **group**, **click any track above or below the selection**. If you **click and move a control element related to one of the tracks removed from the group**, it becomes apparent that the complete group has been cleared.

In order to **remove a track from an existing multi-track selection**, hold down the **"Ctrl" key and click the track name**. If you change a **control element of another grouped track**, it becomes apparent that the tracks are still grouped, even without the track that has been removed.

Alternatively, you may open **the context menu in the track box and select "Track properties > Ungroup track controls"** to clear a group.

Zoom with the scroll bar 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.



A common feature of the preset zoom functions is that they center the play cursor in the visible section, provided it's visible when the zoom command in the section is executed.

Zooming with the mouse wheel

The key combination "**Ctrl**" and the mouse wheel provides simultaneous horizontal and vertical zooming - even during playback.

The key combination "**Ctrl + Alt**" and the mouse wheel provides simultaneous horizontal and vertical zooming.

You can zoom in by rolling the mouse wheel upwards.
You can zoom out by rolling the mouse wheel downwards.

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 play cursor 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 way you can position the play cursor precisely in one go: set the desired playback position roughly by clicking somewhere in the marker bar. . Next, move the mouse downwards while holding down the mouse button to zoom into the project. Correct the mouse position by moving it horizontally, and finally zoom out of the project by moving the mouse upwards.

Deactivate this function at any time via the system settings ("Y -> keyboard, menu & mouse -> Mouse -> Special key performs vertical zoom without shift key").

Zoom Mode

Selecting the magnifying glass icon in the mouse mode bar accesses Zoom Mode (view page 73). Use the right mouse button to zoom out of the project, and click with the left button to zoom into it.

Zoom Tool (Shortcut: Z)

You can temporarily switch to "Zoom" mode by pressing "Z" and then dragging out a lasso with the mouse. By holding down the "Z" key and left clicking, you can zoom into a project in stages; right clicking while holding down "Z" zooms out in stages.

Zooming with the scrollbars

The scrollbars may also be used to change the zoom stages. When the mouse pointer moves over the left and right border of the horizontal scrollbar, it will change into a double arrow. This may be used to move the scrollbar borders, which also changes the length of the section displayed in the window. The length will then be shown on the left of the horizontal scrollbar.

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

Scrolling

The visible section follows the play cursor in so-called Autoscroll Mode during playback.

All commands for setting the play cursor via keyboard or via the menu "Play / Rec > Move play cursor" also move the section with the play cursor. When moving the play cursor with the arrow keys the section will be moved as soon as you leave the window (page mode), with the keyboard shortcut "Alt+Arrow" the section will always be moved in such a way that the play cursor remains in the middle of the VIP window. The "Home" and "End" keys move the play cursor as well as the visible section to the beginning or end of the project.

Should it be the case that you wish to move the visible section without moving the play cursor, 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. This method is very useful for navigating quickly within the arranger together with the vertical zoom (view page 100).

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 the position bar: The red arrow buttons can be used to move the section independently.

With the keyboard: You can set up Keyboard shortcuts for all commands of the menu "View -> Horizontal/Vertical" (view page 816).

Detailed information about creating keyboard shortcuts is available in the chapter "Preset shortcuts".

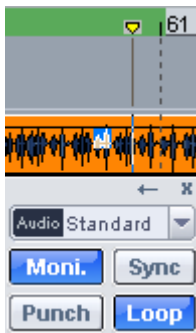
Ranges

Ranges are selected sections of the arrangement that may be selected for editing or access later on. When creating ranges, you are not bound to object borders, individual tracks, or any other limitations like markers. Ranges that you search for are displayed inverted. You can set the step size of the selection by switching on the "Snap/Grid" function in the "Project options (view page 33)".

Working with ranges

Be aware of the following specifics when working with ranges in Samplitude:

- The range remains in "Loop" mode even if the play cursor is repositioned
- In loop mode the entire project is looped, even if you haven't marked a range.
- 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.
- In loop mode the entire project is looped, even if you haven't marked a range.
- Range ends can be adjusted during playback as well.
- The play range can be deleted by dragging to size 0. .
- The current range can be deleted by double clicking the grid toolbar outside of the range.

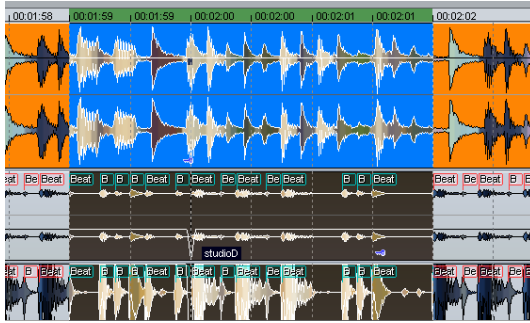


Note: In the general program settings (shortcut: "Y -> Program -> General"), if a check is removed from "Play cursor independent of range", the active range will be defined for the length of the recording or playback process. Samplitude then ceases playback or recording automatically at the end of the active range, provided the "Loop" function is active in the transport controls.

Selecting a range

To select a range, move the mouse pointer to the top half of an object and press the left mouse button. Move the mouse pointer within the object while holding down the mouse button. An inverted rectangle appears between the starting point and the current mouse position. Once you let go of the mouse button, the range will be selected.

The play cursor always automatically remains at the beginning of the range, even if you can't see it at this moment in time. To expand the range to other tracks, click on the top half of the selected object again and drag the mouse down vertically while keeping the mouse button held. Now click the section with the "Shift" key held down, and every object featuring a middle point within the selection rectangle will be selected.



A range may also be selected by dragging the mouse across the timeline. This will then be indicated by a different color. Double clicking this range in the timeline selects a range in the selected track; double clicking again selects the range over all tracks, and double clicking once more restores the simple timeline selection.

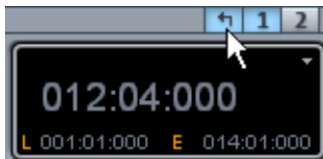
The timeline selection enables the play cursor to be positioned outside the range. The range will remain in loop mode. This way, playback may be started in front of or within a loop. In this case, the range edges may also be modified during playback. The playback range may be deleted by dragging to size 0. . Click the range borders in the timeline to position the play cursor at the playback borders. Double clicking the timeline outside of the range removes the playback 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.

Reactivating a range

Use the shortcut "**Shift + backspace**". By clicking on this command repeatedly you can restore the last five ranges. You can execute the same function by clicking on the left arrow button in the transport control.



Change range borders

Timeline selection enables you to change the range edges by positioning the mouse over them. The mouse pointer turns into a double arrow. The range borders may now be changed by dragging horizontally.

To change the edge of an existing track range (start, end, top edge, or bottom edge), left click inside the range of the existing area and hold down the mouse button. Next, **leave the range in the direction of the edge you would like to change while holding down the mouse button**. As soon as the edge of the existing range has been crossed, the range edge will follow the movements of the mouse pointer. Once the range's edge has been redefined, release the mouse button. The **start of the range** may be changed via the **arrow keys**, and the **range's end** may be changed by pressing "**Shift + arrow**".

Horizontal movement of a range

Left click **within the existing range while holding down "Shift"** and **holding down the mouse button to move the range horizontally**.

Saving and opening ranges / Special range commands

Selected ranges can be saved by going to "Edit" then selecting "Range" from the menu or by pressing "**Alt**" and one of the function keys "**F2-F10**" and opened again with "**Ctrl + 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 as it is used in Sequoia as a source destination edit command. 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 488)".

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

These and further special functions for defining, changing, and using ranges can be found in the "File > Edit > Range" menu. Beneath this you'll find a range editor dialog for the exact numerical input of ranges and the Range Manager (view page 149) for quick and easy viewing of and jumping to ranges.

More information on ranges can be found in the menu reference point under "Menu Edit > Range".

Examples for working with ranges

Example 1: You want to move multiple neighboring objects to a new position. Instead of selecting each object separately while holding down the "Ctrl" key, you can select a range that contains all objects. Select the desired objects via "**Object > Select objects > Select objects under the play cursor/range**".

Example 2: You wish to remove a section of a song from the VIP window completely. This may be, for example, a verse of a song that should be cut out in order to reduce the overall length of the track. Select a range and activate all tracks by double clicking twice to select the affected verse. Then select **Edit > Delete > Delete with time/ripple** to remove the verse.

Example 3: A selected range can be played as a loop. Activate the "Loop" button in the transport console. In this case, the range edges may be modified by dragging the handles on the edges. This enables previewing prior to making a cut.

Markers



Markers can set position points in your project that are of special importance to your arrangement. On opening any set marker, the play cursor will be moved to the defined marker position. Markers are listed along the top line of the virtual project, the so-called marker list, and these may be set when stopped, during playback, or during recording.

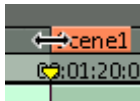
A virtual project in Samplitude can contain any number of markers. The first ten number markers can be saved using the **key combination "Shift + number key"** on the corresponding play cursor position and open them directly via the number keys again.

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



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

In the menu point "Play / Rec > Marker > Marker with name" you may name additional markers freely.



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 them and dragging them to where you want to go, 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 object mode "Group objects", simultaneously moves the marker, independent of the selected track.

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

The marker manager may also be accessed via "View > Manager > Marker manager". This is where you manage and edit the markers you set.

More information about this is available in the chapter "Manager -> Marker manager (view page 147)".

To stretch a range between any two markers, first click on 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" to quickly jump back and forth between the marker positions or the keyboard shortcut "Alt + W/Alt + Q" or select "Play / Rec > Move play cursors > Marker left/right" from the menu.

Alongside the standard markers there are additional markers that can be used in Samplitude as well:

- Markers in wave projects are saved in the audio file (*.wav) as so-called **audio markers** and are available in this form in other applications as well. 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. Audio markers may also be made visible in the view options ("Shift + Tab") in the "Objects" area by marking "Audio markers" with a check mark. 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. **Note:** All time information for the audio marker relates to time positions in the audio material, and not to positions in the virtual project.
- 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** signify a tempo change at a specific position in the project.
- **Beat markers** change the type of beat from the marker position, e. g. 4/4 beat to 3/4.
- **Beat position markers** assign a specific musical position to a specific time position. This way, the bar frame/grid and MIDI events can be easily synchronized with existing audio material.

For more information on Tempo and grid position markers, see "Play/Rec > Marker (view page 301) > Insert Tempo Change/Insert Grid Position Marker" (view page 305) in the menu under reference.

Advanced ruler/time display

Use the grid bar's **context menu to activate a second grid bar** in the arranger and set your own measurement unit (**show 2nd grid bar**). Both grid bar positions can be exchanged (**swap grid**). If you select "**Independent time format**" in the transport window, the upper grid will adapt to a chosen BPM grid, unlike the lower grid. 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 play cursor 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 play cursor (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.

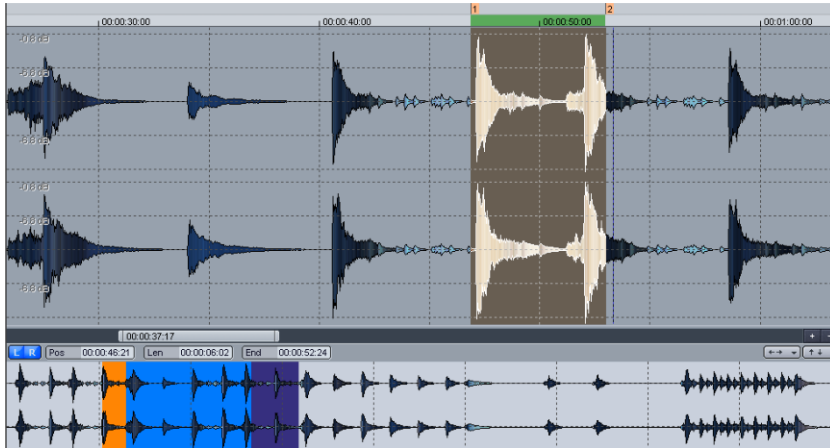
Samplitude as a wave editor

Samplitude executes most tasks completely in the virtual project without having to access the original audio material.

In certain cases, e.g. if audio files are used in other applications, it can be advantageous to **adapt the original material**. In this case, not only the reference objects within a virtual project are edited, but also the wave files (.wav) on your hard disk. When saving the modified files, you can create a copy of the original file to easily access the original file at any time. On the other hand, the command "Object -> Edit a copy of wave content" enables you to open a copy of the wave project for editing purposes.

When using Samplitude as a wave editor, practically the exact same editing options are available as those during non-destructive editing. Real-time effects are not available, since these may only be used on virtual object layers in the VIP, and several mouse modes such as "Universal" mode, "Object" mode, "Curve" mode, "Cut" mode, and "Pitch Shift/Time Stretch" mode.

To open a wave project from within a virtual project, point at the lower half of the reference object and double click while holding down the "Ctrl" and "Shift" keys. Alternatively, you can also select "**Object -> Edit wave project**" from the menu. **A wave project 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.



There are two methods for editing wave projects: "Destructive editing" mode and the virtual "Wave editing" mode.

Destructive Editing

Destructive audio edit mode is the standard mode when the command Edit Wave Project in the Object Menu is used to open wave projects that belong to a VIP object.

Wave projects opened directly via "File -> Open -> HD wave project" are opened in the virtual wave editor by default. This can be adjusted in the program settings. ("Y" -> Program -> General -> Open wave projects in destructive editor mode").

Regardless of the mode you load your wave project in, **you can toggle between destructive and virtual wave editing via "File -> Project Properties ->Destructive Wave Edit Mode"**. If the box is checked, the extension "destructive" will appear in the title list of your project.

In destructive mode wave projects are written directly to the hard drive. Changes in the editing window immediately affect the audio wave 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 wave projects in the program settings ("Y -> Program -> Undo").

Note: At any time, a selected range in a wave project may be transferred from a wave project in destructive mode into a VIP via "Copy" and "Paste".

Non-destructive wave editing

Samplitude also features non-destructive (virtual) wave editing. The advantage of this editing mode is that it works directly in the audio material combined with the features of non-destructive virtual editing. **Compared to destructive wave editing mode, the following differences are apparent:**

The features "Cut", "Copy", "Delete", and "Insert" are non-destructive. The positions where these operations have taken place are marked with dotted lines. These operations may be performed in real time, since it is no longer necessary to copy data to the hard drive for undo functions. All edits are performed when the wave project is saved, and then they are added to the project.

"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.

"Wave editing" mode – master section

If you open the **mixer window** by pressing the "M" key, you will see a minimized mixer on the master channel. This makes **all of the real-time effects familiar from the VIP master section available**, even for non-destructive wave projects. This enables you to apply multiple effects simultaneously when editing waves non-destructively. During destructive editing, however, only one effect may be written to the wave file at a time.

"Wave editing" mode – auto-crossfade

The auto-crossfade option may be activated during any cut or insert operation in non-destructive wave projects

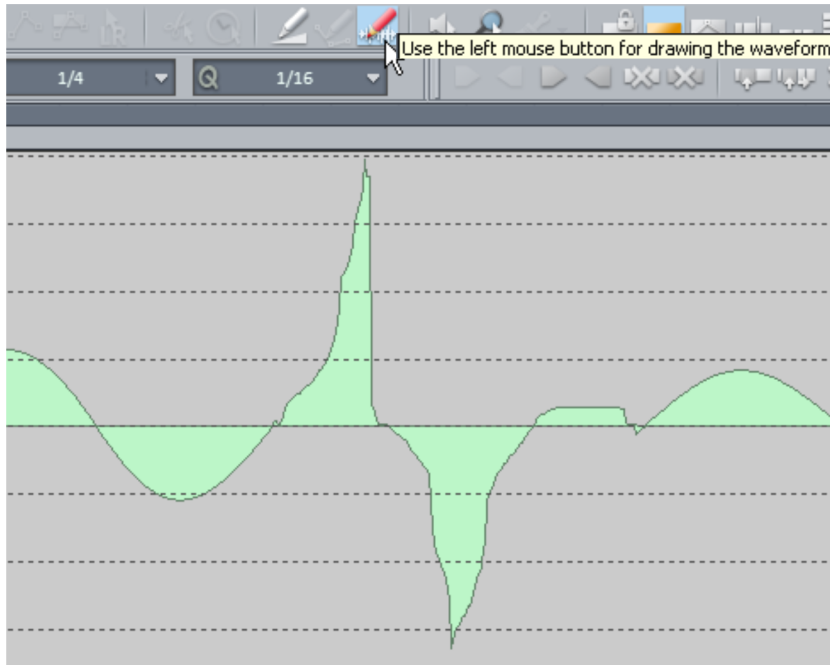
Working with Samplitude – tips & tricks

Here are some hints on how to work more effectively with Samplitude:

New wave project via drag & drop

Click on a selected range from the range list of your wave project and drag this up from the editing window onto a free section of the program background to **create a new wave project**. This enables you to make copies of the entire wave project or just parts of it quickly.

Drawing waveforms with the pencil tool



In "Wave editing (view page 108)" 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.

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 play cursor**. 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 may be executed fastest by clicking on the corresponding keyboard shortcut.
- The **level and panorama curves** may 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 may be moved together.
- **Numerical values** in dialog fields like the display of the playback position in the transport window may 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.

Mixer

- In the mixer, right click on the insert slots **to open** the corresponding **effect dialogs**.
- Double clicking the **knobs/faders** moves them to the **zero position**. Double clicking again restores them to their original position.
- Master normalization in the mixer contains a feature which allows you to quickly **set the output level to 0 dB**. If the limiter in the master section is active, the output level is also monitored to prevent level clipping.
- The mixer function "Mix to file" is located on the bottom right of the mixer. When the "On" button is active, the **master output of the mixer may be written to a wave file or HD wave project during playback**. During playback, change any parameter you want in order to record any live sound adjustments you make.

Recording/playback

- The **recording dialog** may be **accessed** via "Shift + R". Start recording by pressing "A"; "S" ends recording.
- Begin and stop **playback** instantly by pressing the **space bar**. On stop, the play cursor returns to the starting point. . If playback is stopped by pressing "0" on the number pad, the cursor remains at the last position. The exact position may also be moved with the mouse while holding down "0" (scrubbing).

Object-oriented audio editing

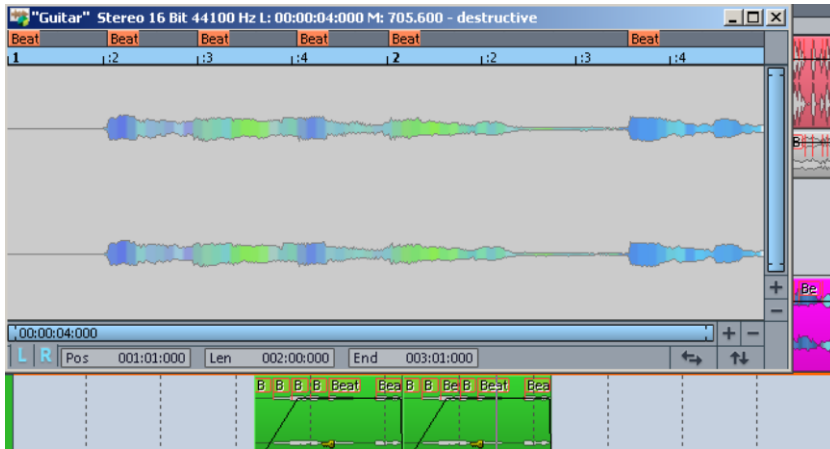
Object-oriented editing describes a method which enables the various ways of editing any selected object, independently of other mixer and track automation settings. This results in a high level of flexibility, even on the object level. You can edit sound material considerably quicker, add various effects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

What is an object?

The object is a unit in Samplitude's arranger window. The object is displayed as a waveform, which may be moved, copied, or cut in any way.

Objects contain references to the basic audio material. You can also speak of referencing an audio file with an object.

An object accesses available audio data and calculates parameter settings like volume, panorama, object length, sound changes, effects, and fades, all in real time. Pitch Shifting, Time Stretching, AUX send feeds, and plug-ins are also calculated.



Object Editor

The centerpiece of object-oriented working is the object editor, which can function as a tool for a single object as well as multiple ones.

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 244)"



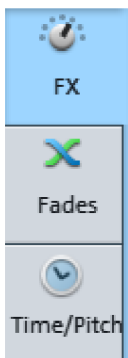
The object editor is divided in three sections "FX", "Fades", and "Time/Pitch". The object editor may be left open to edit objects - the view automatically adjusts to the selected object.

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 50).

The Object Editor can also be opened for an unselected object by positioning the play cursor over the object you want to edit and pressing Ctrl + O. The track containing the object must be activated.

To select all objects at a certain play cursor position, activate a track with no object at that position. Press "Ctrl + O" and all objects located at the cursor position will be selected, the object editor will open for the topmost object in the track list, which also features the lowest track number.

Basic functions: Object editor



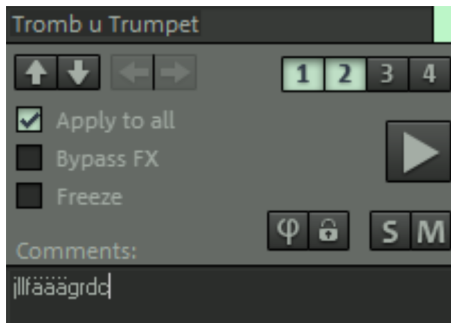
The selection options for the three different dialog windows of the Object Editor are displayed to the left.

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 values.

Time/Pitch: Here you can set Timestretch/Pitchshift algorithms and loops.

The section on the far right is the same for all three dialog windows:



The object name is displayed in the upper text field and this may also be edited.

In the square field to the side you can select the object color.

Using the **arrow keys** you can "jump" up or down to the object of a neighboring track.

Use the **double arrow buttons** to jump to the previous or next object. If multiple objects are selected, these buttons will not be active.

You can save the various object editor settings in the **4 snapshots**. This can be done by "**Shift + left clicking**" on the corresponding button. In case you haven't yet defined 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 the various object editor settings. Also by right clicking on the **numbers keys 1 4** you can load, save and delete snapshots of the object editor's settings.

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, regardless of which of the three object editor window areas, are applied to all other selected objects.

Note:

In an object - **Volume changes** are **transferred relatively**, 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.

FX bypass: this deactivates all real-time object effects, EQ/panorama settings, AUX sends.

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. This way you can bypass the processor-heavy realtime processing of effects.

Play/stop: This button corresponds to the normal playback function (Shortcut: space bar).

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 activated Lock button corresponds to the key button's switch in the 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 settings** (Shortcut "Y") under "Program ->Object lock definitions" (view page 549). Locking can be temporarily cancelled by pressing the "alt" key.

Solo: This button moves the play cursor, during playback, to the start of the selected object and plays that only. Track-dependent buses and AUX paths are also observed at playback.

Mute: This button mutes the selected object. You can also mute the left and right channels individually too.

Commentary: 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 may 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 may 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 via 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.

Here, plug-ins and insert effects can be selected by clicking the **arrow symbols of each plug-in slot**.

Just by clicking you can switch the sought plug-ins on or off. Right clicking opens the interface of the respective plug-in.

The **"FX" button** opens the Effect routing selection dialog (view page 174) for advanced settings and modifications to the plug-in series. Here you can load even more plug-ins than are visible in the slots. To load additional internal effects, you should use the FX/Routing dialog. If you click on the arrow button next to the "FX" button, you can save and load the object effects settings.

The **"Offline-FX"** button opens a selection dialog concerning using offline effects (view page 178). 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. The blue button in the header switches the equalizer on and off. By right clicking on the controller/blue button you can access the dialog window for the fully parametric 4-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 controller opens the "Pan settings" dialog. Use **L <-> R** to toggle between the left and right channels.

Pan: Use the fader on the left to specify the panorama in the stereo picture.

Stereo: Use the fader on the right to set the stereo width.

Balance + stereo enhancer: The pan fader sets the stereo balance. The mid position corresponds to a level of 0dB on both channels. When 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 Panorama+Stereo enhancer: The panorama fader controls the panorama position so that when the levels of both channels are set to the middle, the channels are reduced 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 via 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 like a mixer with mono channels.

2-channel volume: This mode controls the level of the two object stereo tracks separately.

Reset: This button resets the controller and Panorama mode.

Note: For three-times more powerful frequency-selective base width, use the multiband stereo enhancer as an effect insert in the plug-in section.

Vol

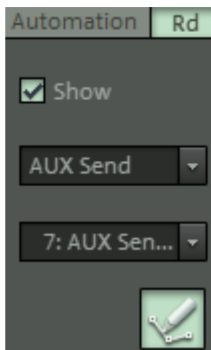
Vol: The volume is controlled using this fader.

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.



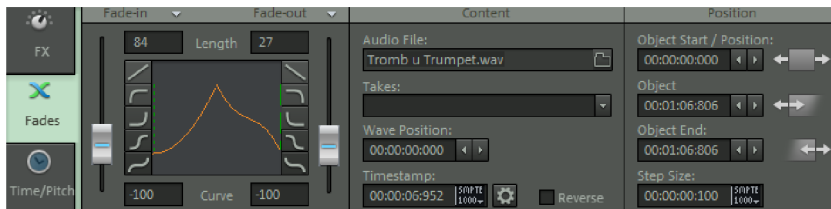
You can make the automation curves of the selected parameter visible in objects with the "View" option.

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

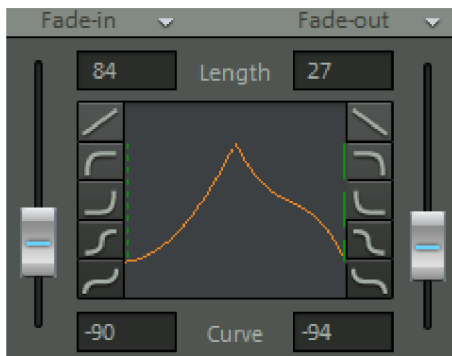
Activate the **Automation drawing mode button** and draw the object automation curve in the corresponding object.

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 may be edited via this dialog.

Using the faders you can adjust the precise settings of the curve shape 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 linear presets of the curve shapes, "Exp" (exponential), "Log" (logarithmic), "Cos" (Cosine) and "Sine". Here you can load separately saved additional curve shapes. 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.

Beneath this you'll find the three "Fade offset" as presets.

"**Fade offset (%)**" defines the percentage of the fade that should extend beyond the original object edge.

- 0%: The fade is located completely within the object
- 50%: the fade is symmetrical to the object edges
- 100%: the fade is completely outside of the original object edge

The object is stretched according to the percentage set in "Fade offset". 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 wave project so that an object may always be faded in or out. For example, if the object starts at the exact beginning of the wave display, you will no longer be able to crossfade as soon as the "Fade offset" rises above 0%.

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

The "**Get global crossfade**" function may also be used. 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 shape for fade-in and fade-out is shown together with the "Fade offset" in the graphic.

Content

Audio file: Here the referenced wave project can be exchanged directly in the Object Editor, copied and renamed. Open the dialog exchange dialog by clicking the folder system beside the name of the audio file. If the audio file is used by multiple objects, you can indicate whether it should be switched for the currently selected object or for 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, they 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 471).

With the option **Reverse playback** an object will be played backwards. The audio files are not changed.

Hint: Please note that there are some limitations when using additional effects on a backwards-playing object. This refers to wave editing or using timestretching effects, especially Elastic Audio. Furthermore, reverse playback requires more resources as track speed cannot be used here.

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.

Timestretching/Pitchshifting



Pitchshifting and timestretching can be executed simultaneously and independently of one another (but not in **"Resampling"** mode).

Pitchshifting: Enter the pitchshift in semitones (mouse wheel + Ctrl) or percentage (mouse wheel + Shift) or as a **factor** of the output material.

Timestretching: Enter the speed correction either as a new object length/**stretch length** or as a **factor** of the output material, or even as the desired speed in BPM (beats per minute).

The button "**<< from loop or range**" makes it possible to extract the BPM value from one selected range or the object length and enter it as the "Original BPM". Enter the number of quarter notes in the dialog that opens and then confirm with "OK". This is important if the object tempo is not entered correctly and another tempo needs to be compensated.

The values are also able to be adjusted in stages using the knob.

If you wish to change the original speed ("Original BPM"), you will see that the speed of the object you wish to correct adapts to that of the set stretch factor. The modes available are "**Resample**", "**Standard**", "**Smoothed**", "**Beat marker stretching (smoothed)**", "**Beat marker slicing**", "**Beat marker stretching**", "**Monophonic voice**", "Universal HQ" and "**True Resynthesis**".

"**Edit**" opens a clear display of the already named parameters. If you move through the individual modes with the mouse, a detailed explanation will appear for each mode.

Detailed information about these algorithms is provided in "Effects -> Time/pitch -> Resampling/Timestretching (view page 647)".

"**Reset**" restores all pitchshifting and timestretching settings. Elastic Audio provides a high-quality tool for pitch customization. The "**Elastic Audio**" button opens the activated object in the Elastic Audio Editor.

The **Use Elastic Audio (Pitch Automation)** , changes the pitch using Elastic Audio.

Loop

In Loop mode (view page 136) **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 play cursor 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 -> Amplitude as a wave editor (view page 108)".

Integrating sections from wave projects as objects

To integrate a section of a wave project into a virtual project (VIP) as an object, open a wave project in an empty project window and drag the range up that should be integrated into the virtual project (VIP). Then open a virtual project (VIP).

Click anywhere in the VIP in the upper half of the desired track at the position that should display the beginning of the new object. The play cursor now moves to the position where the object will be inserted. The track is selected. Click **"Object -> New object"** to insert the previously selected range.

Alternately, press **"Return"** to display the VIP and wave projects one under the other.

Click the selected range of the range list in your wave project. Now drag this section of the wave project up into the desired track of the VIP. The selected range is now available as a new object in the VIP.

The difference between loading and importing audio files

Unlike during loading, the selected audio file is converted to .wav format when imported and then copied to the hard disk. This is only necessary for file formats that cannot be opened directly using Amplitude, e.g. Windows Media Files and certain MPEG files.

Increase the performance of your computer by importing compressed formats like MP3 or OGG-Vorbis instead of simply loading them to bypass real-time decoding.

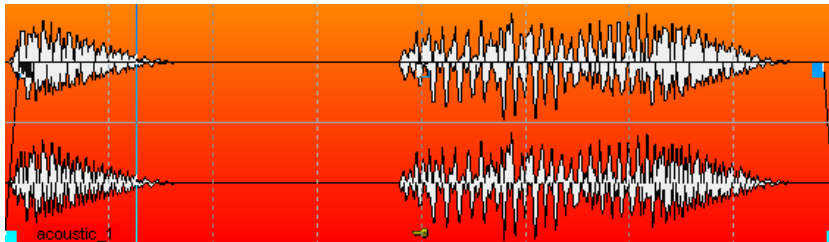
Integrating CD tracks as objects into VIP

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

- Click anywhere in the upper half of the desired track at the position that should display the beginning of the CD track. The play cursor now moves to the position where the CD track will be inserted. The track is selected.
- In "**Menu CD/DVD**" -> **Load 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

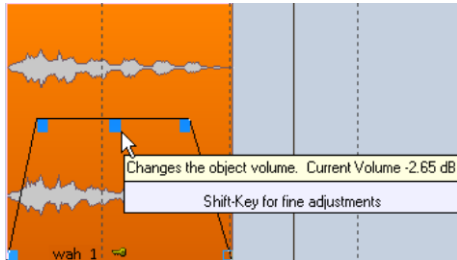
Each object may be selected with the mouse. To do this, click the lower half of the object with the left mouse button. The selected object will be recognizable by a different background color and the five handles that appear on its corners and in the center of the object.



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

Object handles

When 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 wave project. Use the object editor to change the length as well.

The length of an object depends on the audio file that is accessing the object. The longer the audio file is, the longer the object can be. The referenced audio file restricts the length of the object: the object may not be longer than the audio file.

The **fade handles** (top left and top right) are used to fade an object in or out, and different curve shapes are possible. Use the object editor to adjust the curve shape precisely.

The **volume handle** is at the top center of the object. You can use this to adjust the volume. The volume level is displayed above the tool tips.

The object name is visible next to the left length handle on the right hand side and the key symbol to its right may be used for locking objects (view page 130).

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

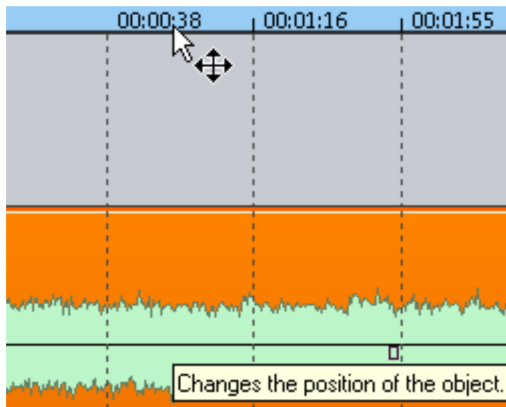
Objects may be selected in various ways. The selection may be expanded, reduced, or reversed at any time.

- 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..

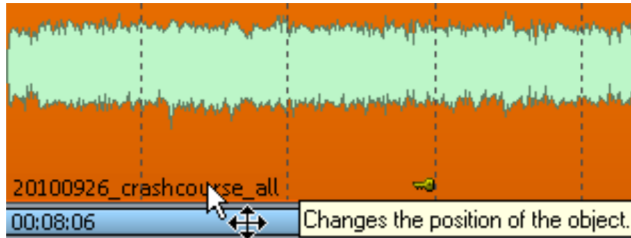
- The current object selection may be reversed (invert). To do this, select **"Object > Select objects > Switch selection"**.

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 move an object in 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 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"**.
- **To move 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 may 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 may be moved to the tracks below.



nudging objects

- "**Object -> Move objects > Object step 1/2**" features a number of commands for specifying individual keyboard shortcuts in the system options.
- To change the step widths 1 and 2, access "**Object -> Move objects -> Object/Fade Step Settings**".
- Alternately, select the objects you wish to move and open the Object Editor. In the Position/Fades window 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.

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.

Duplicating, copying, and inserting objects

Duplicating one or more objects is done by moving objects while holding down the "Ctrl" key.

Copying and inserting objects is done in the familiar manner with the keyboard shortcuts "Ctrl + C" and "Ctrl + V".

Inserting an object with ripple

The keyboard shortcut "Ctrl + Alt + V" inserts a previously copied object at the play cursor position; the objects following this in the arranger will be moved back correspondingly ("ripple").

Mute objects

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



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

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

Change waveform view

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

Detailed information about the display options is available via the menu reference under "View -> VIP display (view page 491)".

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 play cursor by pressing the "Ctrl" key and simultaneously scrolling with the mouse wheel.

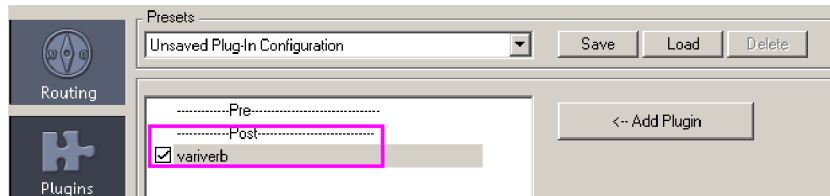
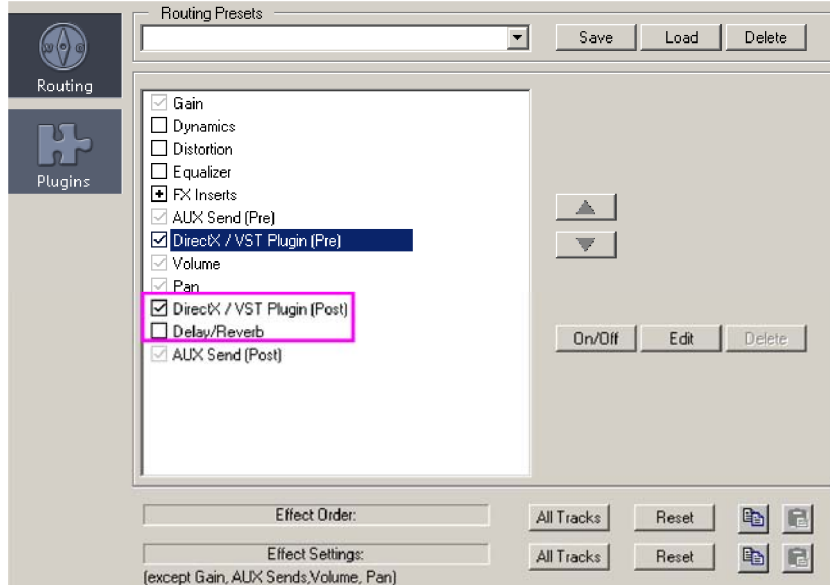
Edit fades in the object editor or by using the handles

To quickly produce a fade, click and drag the fade handle of the selected object. The curve shape that was set in the object editor is used in this case.

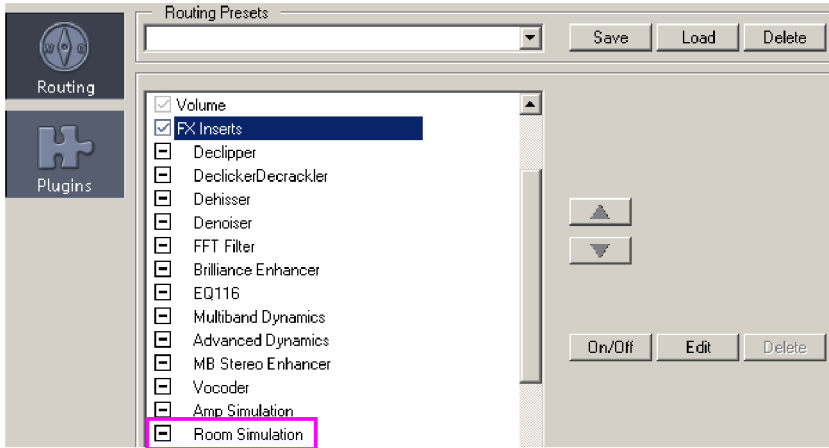
The object editor provides a selection of fade curves when the "Menu" button is pressed. Select the fade lengths in the field beside this.

Reverb in the object

Reverb times fade away if the **Reverb effect in the routing is situated after the "volume"**. For standard routing, only the post DirectX/VST plug-ins or internal delay effects apply.

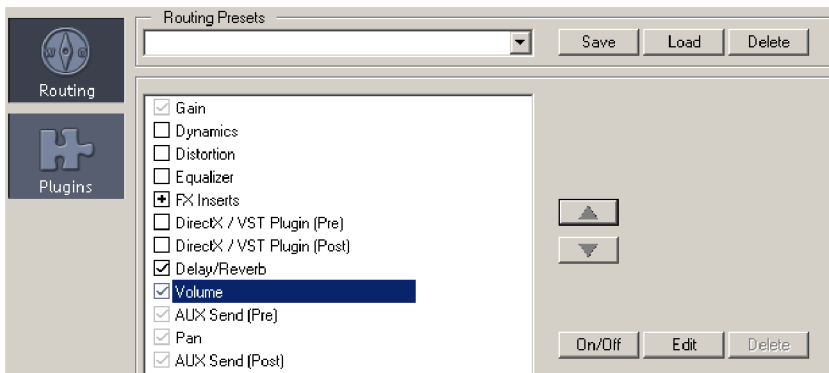


For reverb times with RoomSim change the effects sequence so that "Volume" is placed before the "FX inserts".



The reverb time depends on the option "**Maximum reverb time for objects without fadeout (sec.)**", which is located in the playback options (view page 578).

To ensure that reverb does not continue past the fade out, but rather that it breaks off, arrange the object's **parameter "Volume" in the FX/routing dialog after the effect** (FX inserts or DirectX/VST plug-in (pre) or DirectX/VST plug-in (post)).



Locking objects



Just click the **key symbol** on the lower border of each object to protect an object against accidental moving. This function corresponds with the "Lock" command in the object editor.



Clicking on the key symbol again unlocks the object.

Multiple selected objects may be locked with a mouse click.

The menu entry "**Object -> Lock objects -> Lock objects**" applies to single and multiple objects.

If you wish to lock all objects of a track, click on the lock symbol in the track overview box or Track Editor.

Locking against fades, changes to length, or deleting may be set via "**Object -> Lock objects -> Lock definitions**".

Temporarily disable the lock function by holding down the "Alt" key while clicking objects.

Editing an object destructively

By editing an object destructively, to add complicated effects to the audio material, severe loads on the CPU may be reduced during playback in the virtual project (VIP).

To edit an object destructively, right click the object to access its context menu and select "**Wave Editing**". The wave project will open. The wave may now be edited as desired. Samplitude now functions as a wave editor (view page 108).

If an audio file that is contained in multiple VIP objects is edited in destructive mode via "**Object -> Wave Editing**", then all of the associated objects in the VIP will access this audio file. For example, if reverb is added to an audio file in destructive mode, all objects that reference this audio file will now feature reverb.

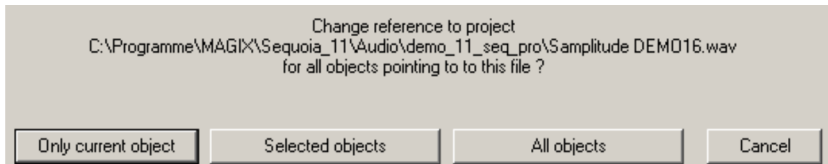
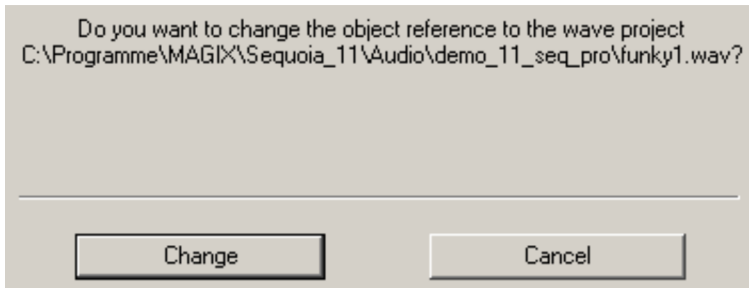
On the other hand, to edit only certain objects destructively, select the command "**Object -> Edit a copy of wave content**". Samplitude will produce a copy of the audio material and add this to the project folder. The selected objects will now refer to the copy that is created. Following this, destructive effects may be applied to the selected objects without influencing other objects.

The button "**Advanced options** (view page 601)" in the corresponding effects dialog provides access to saving options for the copy.

The take manager (view page 150) provides an overview of the versions of destructively edited files and select them for the respective object.

Markers		Ranges		Takes	
⚙️		Take Composer			
File	Recorded		T		
Samplitude...	13.09.2010	12:27:35	01		
Samplitude...	13.09.2010	12:29:26	01		
Samplitude...	29.09.2010	11:36:17	25		

In order to reference audio files for other objects in the VIP retroactively, click the folder symbol in the object editor to the right of the audio file. One of the following dialogs will appear:



Select whether the file should be swapped for the current object, for all selected objects, or for all objects that reference the file to be swapped.

The option "**Create copy**" is activated by default for all destructive effects for safety. 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 destructive editing.

Note: Temporary files for the "Undo" function are only created with destructive 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 play cursor, right click on the object you want to move. In the context menu that opens select: **"Object -> Edit/move crossfade/edit -> Object to play cursor position"**.

Group/ungroup objects

Select all objects for the group.

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

Click the "Ungroup" symbol in the program toolbar or right click one of the selected objects and select **"Object -> Ungroup objects"** from the context menu to clear the group.

"Object -> Select objects -> Temporary 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 repeatedly using the function, the removed object is moved back into the group.

Shortcuts: Ctrl + Shift + U
 Shift + Ungroup button

"Object -> Select objects -> Temporary exclude all objects from groups" 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, then the groups will be restored and the button will stop blinking and return to inactive status.

Shortcuts: Shortcut: "Shift + Alt + Ungroup"

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

If **"Group color"** is activated in the display options, a different color will be assigned to each group. This helps to easily distinguish the groups from one another. You can also activate display of the **group numbers** in the VIP display. Individual objects may be removed from groups by pressing **ungroup** while holding the "Shift" key.

Detailed information about the display options can be found in the menu reference under "View -> VIP display (view page 491)".

To learn more about grouping tracks (multi-track selection), see "Working in the project window -> Selecting/grouping multiple tracks (view page 98)".

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

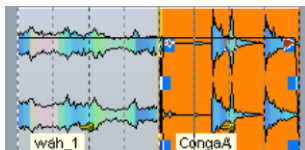
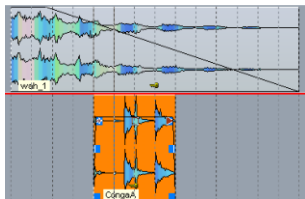
Only one object can be played per track (channel) at any given time.

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.

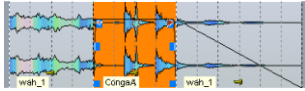
If "**Auto Crossfade**" is active, a crossfade will be created automatically as soon as the overlapping object is faded in.

Overlapping two objects via the keyboard shortcut "**Ctrl + V**"

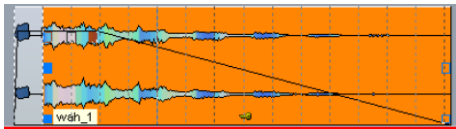
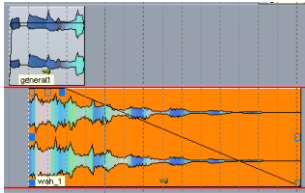
If you drag a shorter object completely into a longer one, then the shorter object will replace the longer object from the start of the shorter object.



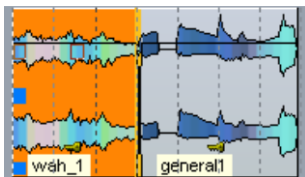
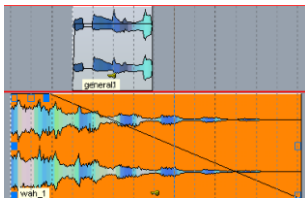
If you hold the keyboard shortcut "**Ctrl + V**" before releasing, then the shorter object will also replace the longer object across the entire length of the shorter object, but the end of the longer object that extends past will remain.



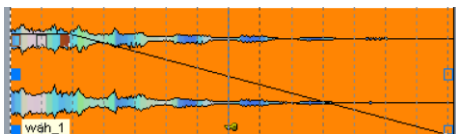
Now try it the other way around and drag a longer object over a shorter object; the shorter object will be replaced from the start position of the longer object if its start position lies behind that of the shorter object.



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 wish 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 122). 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.

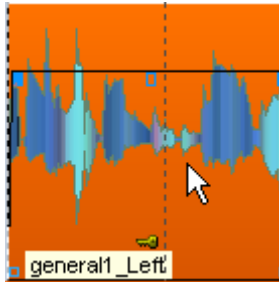
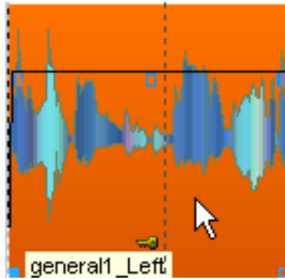
Replacing an audio file below the object



Open the object editor and switch to the view "Position/fades" or to "Max" view.
Now click the folder symbol in the object editor to the right of the name of audio file. A new wave file may be selected in the following dialog.

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. Of course, 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 then 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.

Manager

The managers combines frequently required management and control functions within one window, such as for markers, tracks and CD tracks. This includes a file browser for finding, previewing and loading media files.

The manager may be opened inside or outside of the Docking bay (view page 50). The manager window is multi-monitor enabled and saves the last position when closing. Multiple managers in various different views can be opened simultaneously and independently of one another.

Manager	Shortcut
• 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
• VSTi-Manager	Ctrl + Shift + I
• Routing manager	Ctrl + Shift + Alt + R

To open the manager, click "Manager" in the toolbar or "Manager" in the "View" menu. The menu item "Open new manager" also allows you to open additional manager windows.

Drag the bar above the manager tab upwards to enlarge the manager window. Double click the bar to uncouple the manager window.

Note: Shortcuts work with already opened managers and change to the view which was assigned to the shortcut. If several manager windows are open, the shortcut affects the first opened or currently active window.

By right clicking on the individual columns of the respective Manager window further editing options will be available.

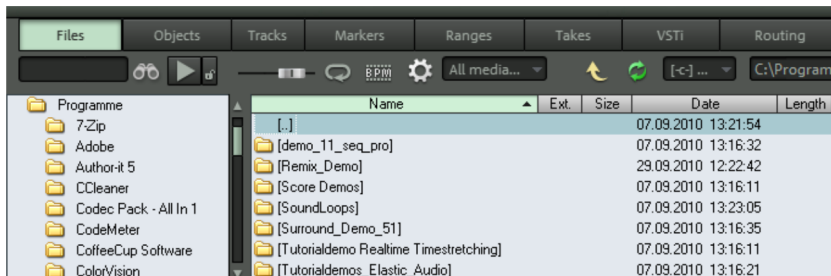
File browser

Samplitude contains a file manager for previewing and loading files directly into projects. The browser may be used to create favorites folders and provides access to all recently opened folders.

To display the file manager, click the "Manager" button in the toolbar, and then select the "Files" tab in the manager window.

Menu: Tools -> 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:

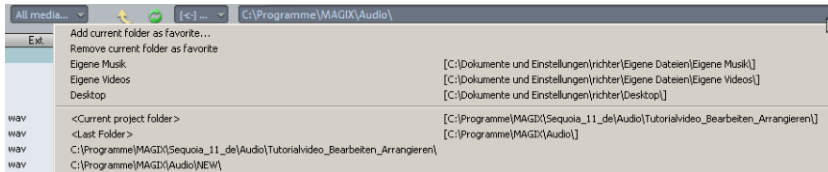
- Name
- Extension
- Size
- Date
- Length
- Resolution in bits
- Stereo/mono
- Sample rate
- Timestamp
- Description/title
- Originator/artist
- Path
- BPM: Specified by the loop length

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 may be previewed with the playback symbol in the file manager. Any highlighted audio file is played immediately if the "Autoplay" box is checked. A complete list may be previewed simply by using the cursor keys. Clicking the play button again stops continuous playback. The **fader** symbol may be used to control the volume.



Playback does not proceed via the mixer or internal effects, but rather via the global playback device selected in the playback parameters (keyboard shortcut: "P"). If no playback occurs, check to ensure that the routing settings for the device are correct.



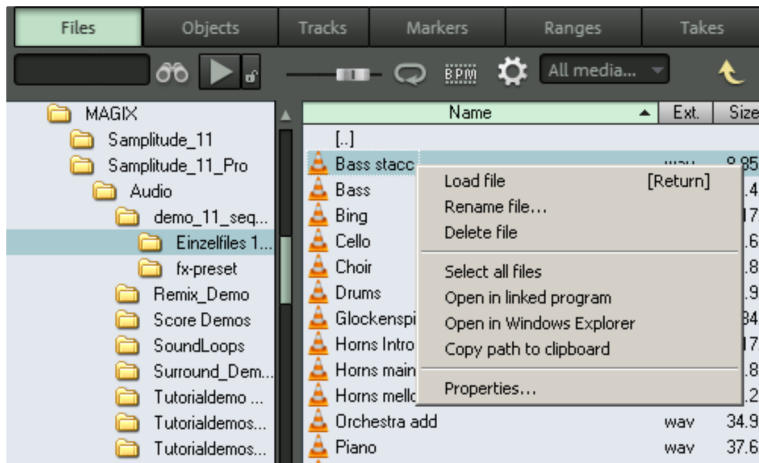
BPM sync: This option enables the selected loop to be previewed in the VIP's tempo. The original BPM is specified in this case by the length of the file. When playback is running, loops are positioned at the beginning of the beat by double clicking and then added into the arrangement as loop objects.



Looped preview enables the selected file to be previewed in a loop.

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

Load files from the file manager into projects



Media files may be inserted into an opened project at the desired play cursor position via drag & drop or by double clicking. If the file browser list is in the foreground, press "Enter" for the same effect. Right clicking the desired file opens a context menu. This enables you to open, rename, and delete files and folders. This also provides access to the functions "Select all files", "Open via standard link", "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.

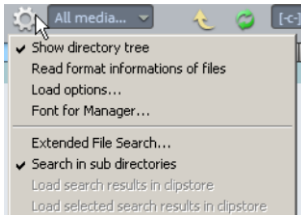
Open file in wave project view: Drag an audio file from the list and drop it into an empty space or into the title bar of the program window.

If several files are selected in the list, all files will be inserted at the current cursor position as new objects, and the loading options ("Options") will also be applied. This allows you to 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 via the program's "Load" dialog.

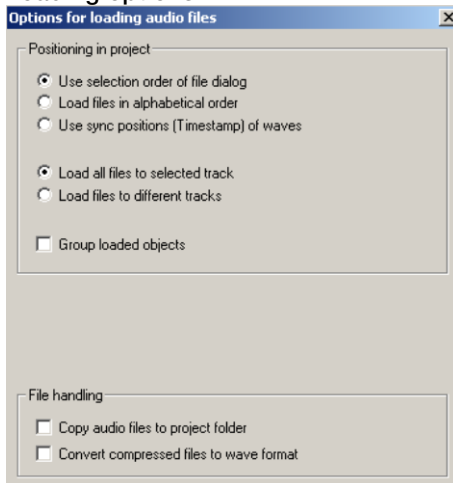
File manager options



Additional options may be accessed via 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 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.

Group loaded objects: All loaded files are grouped. They can be ungrouped anytime.

File management

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 may also be loaded and played directly by Samplitude. This does, however, require increased CPU calculation. Activate this option to convert files like this into an uncompressed wave format.

Extended file search

Additional criteria are available for extended file searching, e.g. **Date of last file change between**, **Date of file creation between**, **Files with audio length between**, and **Files with sample rate of**, **File name**, **Project comment**, **Description/title**, **Originator/artist**:

The screenshot shows the 'Extended File Search' dialog box with the following settings:

- Date of last file change between: 08.07.2008 and 08.07.2009
- Date of file creation between: 08.07.2008 and 08.07.2009
- Files with audio length between: 000:00:000 and 1800:00:000
- Files with samplerate of: 44100 Hz
- File name: trumpet
- Project comment: (empty)
- Description / Title: trace
- Originator / Artist: (empty)

Buttons: Start search, Close

Note: Deleted files will always be removed to the recycling bin. "Shift + Del" deletes directly without placing the file into the recycling bin (after confirmation request).

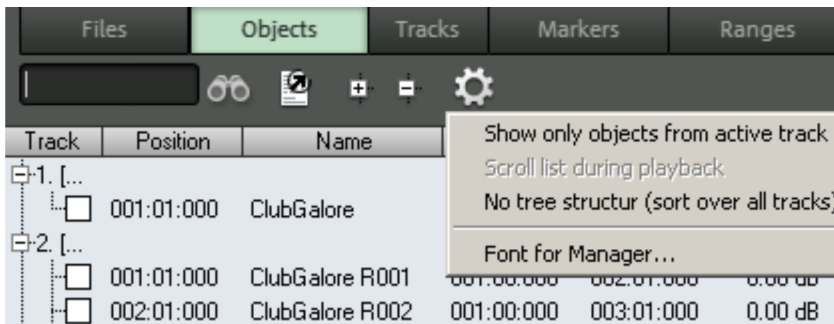
Object manager

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, click the "Manager" button in the toolbar, and then select the "Objects" tab in the manager window.

Menu: Tools -> 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 marker manager list. The following information will be saved:

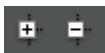
- Project name and project path
- Track and object name
- Start position in the project
- Source file path

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

Object view and selection

These objects are displayed via 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.

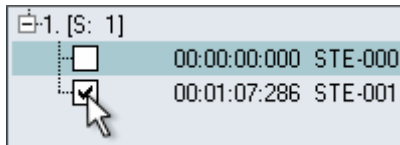
You may optionally choose to display objects in the current tracks only (toolbar: "Options").



You can use the +/- keys on the toolbar to open and close the track object display for the complete window.

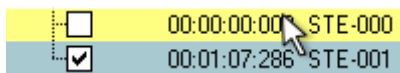


Click on the "+" symbol left of the track number to explode the view of a single track.



Select objects: A square check box in front of the object name enables select of a single 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 via the number of the object group in the "Group" column.



Objects that are currently playing are highlighted with color in the object manager.

Searching for objects: Like in other manager windows, the object manager also provides a search option that allows you to search for objects in the current window. Enter a search item into the box and press "Return". 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 via its context menu. You can rename Objects by double-clicking on the Object names and entering a new name. Alternatively 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 may edit the following parameters via the object manager:

- Start position
- Name
- Length:
- end cap
- Volume
- group
- Lock

To edit a parameter, double click the respective entry and enter a different value. Numeric values may be changed by dragging with the mouse, and "Ctrl" and "Shift" allow larger or smaller changes. "Tab" moves to the next editable value. "Play cursor up/down" navigates vertically within a column, provided this is a text column.

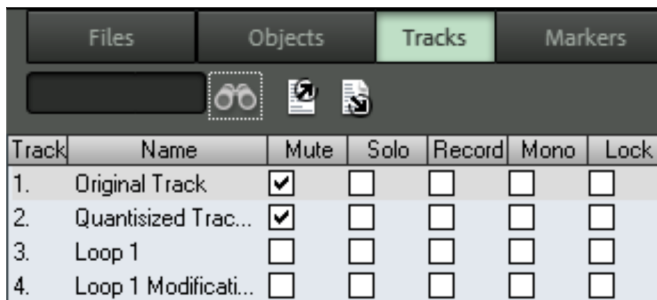
Track manager

All 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 view the track manager, click on the "Manager" button in the toolbar and then select the "Track manager" tab at the bottom of the manager window.

Tools -> Manager -> Track manager

Keyboard shortcut: Ctrl + Shift + S



Track view and selection

The track manager contains all tracks of the current project. Double click the track number to select a track. The track will be displayed in the project window. If the track is outside of the visible display, the project window will scroll to this vertical position.

Search for tracks: Like 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 item into the box and press "Enter". Located tracks are highlighted.

Sort, rename, delete, and insert tracks

To sort tracks in the track manager, just pick them up with the mouse and drag them vertically to the desired position.

Delete tracks via the manager by selecting one or several of these tracks and pressing the "Del" key. Alternatively, you may also use the context menu to delete a track.

Rename tracks by double clicking the track names and entering a new name. Alternatively, the context menu also provides an option for this. "Tab" moves to the next editable value. "Cursor up/down" navigates vertically within a column, provided these are text boxes.

You may also insert new tracks directly in the track manager. Right click on a track entry and select "Insert track" from the context menu. You may also add track folders, submix buses, AUX buses, Surround buses, or attachments. The "Track settings" are also made available by right clicking.

Hiding and displaying tracks

Besides the "Solo", "Mute", and "Record" features, you can also hide tracks. To remove a track from the project window or mixer, click the check the box in the "Arranger" or "Mixer" column. Hidden tracks are still available and are still played and edited while they are hidden.

To display a hidden track, reactive the check box.

Write 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 this imports a list of track names from a text file.

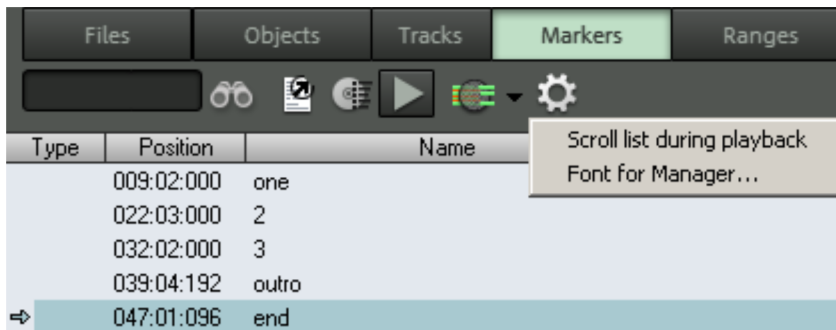
Marker manager

The marker manager displays all markers contained in the current project (regardless of type), 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 then select the "Markers" tab in the "Manager" window.

Menu: Tools -> Manager -> Marker manager

Keyboard shortcut: Ctrl + Shift + Alt + M



The following marker types may 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 may be exported as a text file. 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 the file in the project file (Projectname.txt).

Marker view and selection

The marker manager displays all of the current project's markers.

Marker filter: An optional marker filter limits the view to selected marker types. Select all of the marker types you want to display in the filter menu by placing check marks and then clicking the filter button. All non-selected marker types are hidden.

Find markers: Like 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 item into the box and press "Enter". Located markers are highlighted.

Set range between selected markers: Set a range between the selected markers via the context menu.

Jumping to and playing from marker positions

Double click a marker in the type column to position the play cursor there.

Playing markers: Select a marker and click on the "Play marker" button in the toolbar. A range surrounding the selected marker is created and playback begins.

Deleting markers, changing a marker's name, position, and type

Delete objects directly in the object manager by selecting one or several objects simultaneously and pressing the "Del" key. Alternatively, you may also use the context menu to delete a track.

Rename markers by double clicking the marker name and entering a new name, or use the context menu. Alternatively, the context menu also provides an option for this. Use the same method to change a marker position directly from the list. The "Tab" button switches to the next editable value. "Cursor 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.

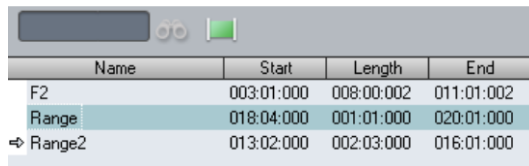
Range manager

The range manager displays all areas contained in the current project and makes it possible to select them directly from the list.

To view the range manager, click the "Manager" button in the toolbar and then select the "Ranges" tab at the bottom of the manager window.

Menu: Tools -> Manager -> Range manager

Shortcut: Ctrl + Shift + Alt + B



Name	Start	Length	End
F2	003:01:000	008:00:002	011:01:002
Range	018:04:000	001:01:000	020:01:000
↔ Range2	013:02:000	002:03:000	016:01:000

Defining and searching for ranges

Defining ranges: To save a range in the manager, define a range in the project window first. 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 with the indicators F2, F3, etc.

Search for ranges: Like in other manager windows, the track manager also provides a search option that allows you to search for ranges in the current window. Enter a search item into the box and press "Enter". Located ranges are highlighted.

Edit range parameters

Edit the following parameters via the range manager:

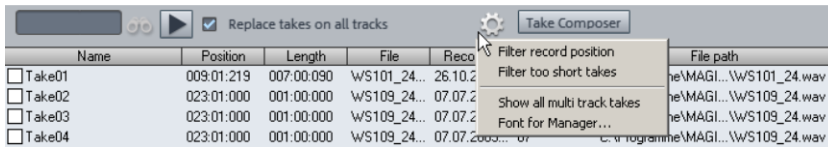
- Name
- Start
- Length
- End

To edit a parameter, double click the respective entry and enter a different value. Numeric values may be changed via drag & drop; "Ctrl + Shift" enables larger or smaller value changes. "Tab" moves to the next editable value. "Arrow up/down" navigates vertically within a text column, provided these are text fields.

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 86), and destructive object edits (view page 131).

When graphical comping with the take composer (view page 152), 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 multi-track 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.

Examples for using the take manager:

- To select the best takes after some punch loop recording sequences
- To find the best takes from several recording sequences using predefined bar positions
- For a well-structured display of all available takes in a specific SMPTE period
- To merge the best parts of existing takes of recorded vocals to 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 is identifiable via the check mark in front of the name in the take list. All the takes matching the selected object are shown in the list. In the basic settings, this refers to all the takes from the same track and original position.

Take manager - options

Filter recording position: Only takes that overlap the same original time positions are shown, e.g. the selected object.

Filter out takes that are too short: Only takes that are at least as long as the currently selected take are shown.

Show all tracks: Takes of all tracks will be displayed.

Manager font: Use the dialog to specify the font, format, and size for the take manager

Select takes

To select a take for the object, place a check mark before the desired take, or press "Enter" after you have selected the corresponding take in the take manager.

Select a take directly in the arranger window by applying "**Ctrl + right click**" to the object.

All of the takes shown in the take manager and the currently selected take are arranged one below the other in the **take composer**. Takes may be compared directly in the take composer, and a combination of the various takes may also be cut out.

Rename take: Right click on the take to open the context menu and rename it, edit the recording position, or delete the take. Multiple selection is also possible.

Edit recording position: This option enables you to make changes in the take manager's position field.

Delete takes: Right click on the take to open the context menu to 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: In case the audio file is no longer referenced via objects, but contains various takes, then not every take has to be deleted in order to fully remove it from the hard drive. Instead, use the "Remove unused samples (view page 465)" function from the Tools menu.

Note: For a better overview, easy-to-use take-naming options are available in the recording window (view page 582) even during recording.

Take management for multi-track recordings

For multi-track 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 multi-track recording to be exchanged in one step.

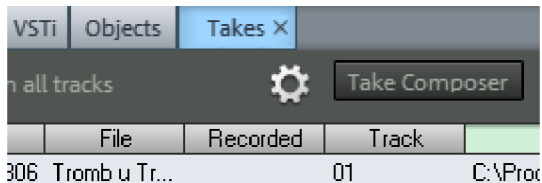
Note: Since the take manager display refers to the last clicked object, the reference track may be switched at any time.

Take Composer

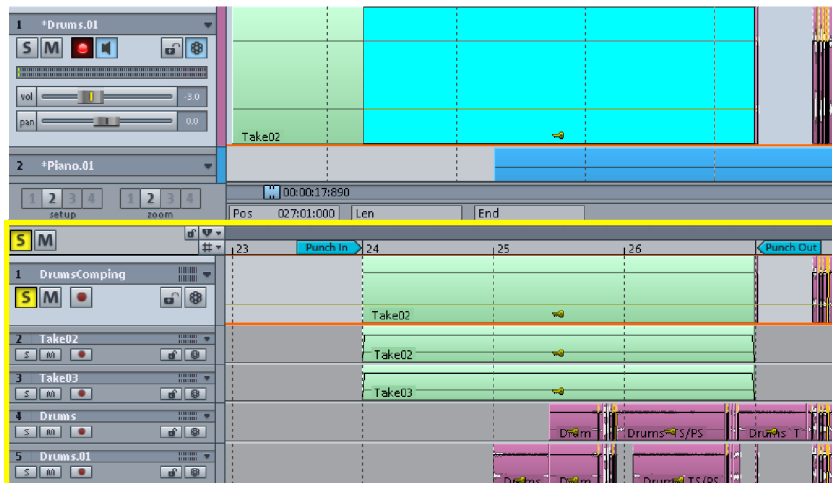
The take composer is an editing window that displays a **selected track's** individual **object recording runs** (takes) and the created **Revolver Tracks** (view page 62) below each other. Produced Revolver Tracks are located below the takes.

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 manager area.

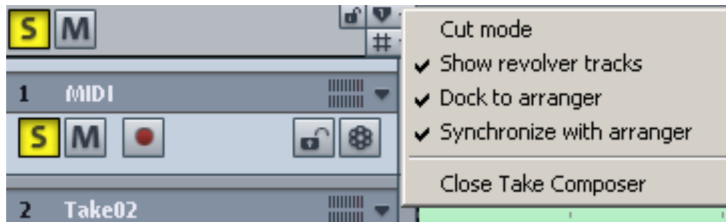


The take composer opens as an extra window below the arranger.



In the take composer use the space bar to initiate playback in sync with the arranger.

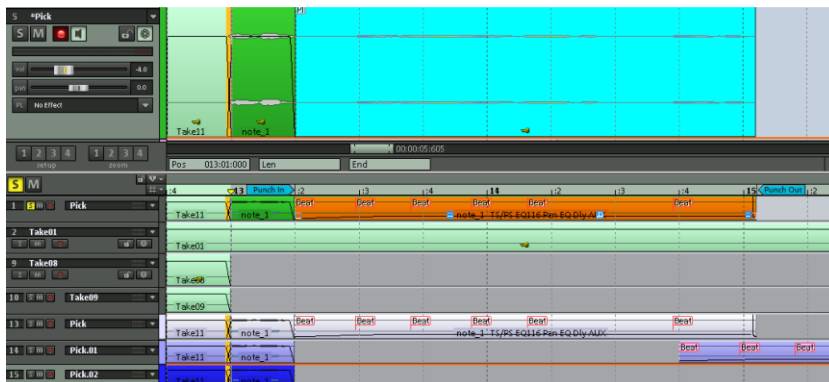
In the manager's marker menu you can specify, if you want to display Revolvertracks, if the take composer should open in its own window or should be docked to the arranger, and if the take manager should run in sync with the arranger.



To quickly execute **comping** (see below), switch to "Cut" mode or close the take composer.

The current take selection of the respective track is displayed in the first track's take composer. During so-called "**comping**", you can edit the best passages of the individual takes together in this master track by using the editing operations described below.

The individual recording takes are arranged in the take composer one above the next starting from track two.



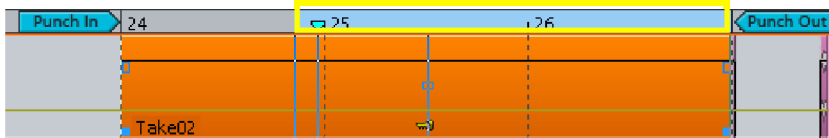
Monitoring with the arrangement

The take will be played together with the starting arrangement when playback is started with the **space bar**.

Monitoring without the arrangement

In order to play the take without the starting arrangement, start playback with the **Play** button on the transport console or via "**Ctrl + space bar**".

A range will also be created that stretches from the cursor position across all selected objects.



To listen to the individual receiving passages switch the corresponding track to "Solo" and start playing. The default solo modus is "Exclusive solo".

In case multiple takes or tracks should be played back simultaneously in the take composer, click the corresponding track numbers and then press the shortcut "Alt + S" in the corresponding track header for the desired tracks. You can switch all selected tracks to "Solo" mode.

Note: If you have an object selected in a take, then the key combination "**Ctrl + space bar**" has the effect that you will only hear this object during playback – even if "Solo" is active on another track.

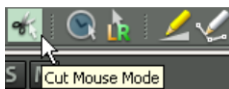
Comping

"Comping" refers to the process of combining several takes to create a single one.

Comping in cut mode

For this process, the take composer offers a **specially modified scissors tool** for quickly cutting/copying take sections into the first take composer track.

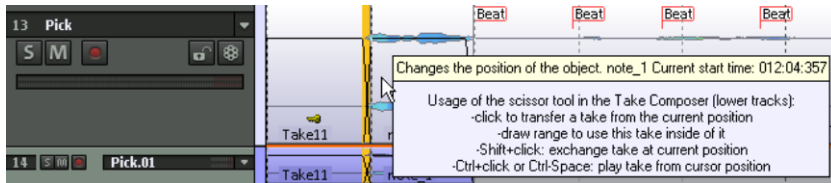
Select the scissors tool by selecting "Cut" mouse mode. The mouse pointer becomes a scissors icon.



Now click on the desired cut point in the respective take. The selected take is inserted from the current position up to the end of the first track in the take composer. Auto crossfade (view page 526) mode is activated as a default during the process. Cut the suitable sections from the respective take in this way along the timeline into the top track.

If you only want to transfer specific ranges of a take into the comping track, drag out a limited range with the scissors tool by holding down the mouse button. When the mouse button is released, the section indicated in the respective take will be copied to the first track.

The keyboard combination **"Shift + click"** replaces an object in the comping track with the contents of the take composer track over which you are holding the scissors tool.

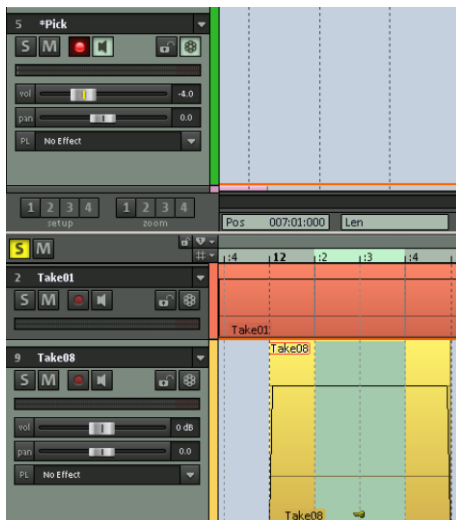


To play a take from the cursor position in "Cut" mode, use the keyboard shortcut "Ctrl + mouse click".

Comping in "Universal" mode or in "Range" mode

Another comping option is to select the desired take range in a track.

In "Universal" or "Range" mode, drag out a range in a take composer track. Use the keyboard shortcut "Shift + C" to transfer the selected object's section to the first track. Auto crossfade mode (view page 526) is also activated by default during this process.



If however you don't mark/select a take range, the shortcut "Shift + C" will insert the selected take from the current position up to the end of the first track in the take composer.

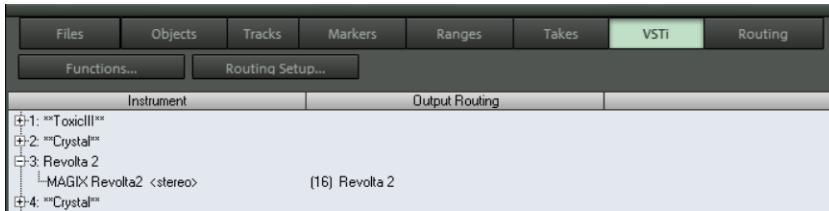
This enables you to select the best parts of the available takes and add them to the top track in the take composer. These parts may be combined to create the perfect take.

When the take composer project is closed, changes may be confirmed, which transfers the take manager's comping track into the original project.

VSTi manager

The VSTi manager facilitates easy display, insertion, and deletion of MAGIX, VST, and ReWire instruments in the current project. Right clicking the selected instrument opens the plug-in dialog for the selected VSTi. If the plus sign beside the instruments display is clicked, the outputs to the assigned tracks will be shown. By right clicking "Output to track", a track may be selected which should only be fed the individual output of each. The assigned output then appears in the target track's plug-in slot.

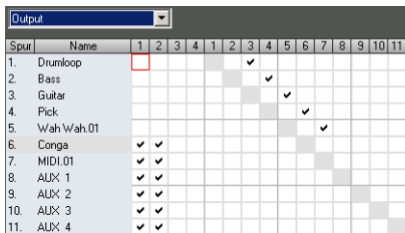
The general instrument outputs are routed in the VSTi manager via **"Routing setup"**. Specify how the outputs should be routed in the dialog that appears. Either route all outputs to the current track or create new mono or stereo tracks for each output. Pressing the "Del" key removes a selected instrument completely from the project. If only individual output signals were selected, routing to a track is canceled by pressing the "Del" key.



Detailed information about instrument routing settings and about managing single outputs is provided in the chapter "Software/VST instruments -> "Routing instruments (view page 318)".

Routing manager

The routing manager provides a clear overview of the matrix of inputs and outputs and AUX connections of all tracks.



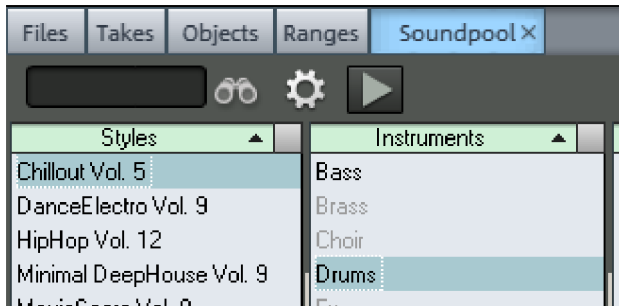
You can assign several tracks to the same inputs and outputs by selecting the desired I/O for the first track and then assign the same I/O by pressing "Shift" + clicking on the last track. Routing may be arranged vertically or diagonally.

In the output section, all tracks are displayed as potential submix busses behind the playback devices; a track may be quickly converted into a submix bus by clicking the corresponding matrix field.

Similarly, an AUX bus may be routed to an individual track. Right clicking allows you to define the AUX bus as a pre-fader send or as a sidechain send, or access the stereo editor for panorama specification for the respective AUX bus.

Soundpool manager

In the Soundpool Manager you can integrate you Soundpool loop content in a clear form.



In the input field next to the magnifying glass enter a combination of letters which should be in the loop you're looking for. Clicking on the magnifying glass symbol starts a search for the corresponding loop name.

You can access a selection menu by clicking on the cog symbol. Here you can:

- Add styles: Search the folder system for the desires soundpool style
- automatically play the selected loops
- Hide instruments without available loops
- reset the soundpool. The list of styles shown in the Soundpool will be deleted.
- Clean Soundpool: With this function only the styles available on the hard drive will be displayed in the Soundpool. Soundpools on storage devices as well as styles deleted or moved from the hard drive will be removed from the Soundpool view.

With the play arrow symbol you can start and stop the selected loop.

Mixer

Mixer overview

You can open the Samplitude mixer via "Window > Mixer". The mixer is very similar to a hardware mixing board. As a result of the digital environment, however, Samplitude's mixer is far more flexible than an analog mixer.

All settings made in the mixer are processed and edited in real time. This includes track and master effects, panorama and level settings, device assignments for individual tracks/buses, and device 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.

Alternative mixer skins

The system options (shortcut: "Y") in "Design -> Skins" feature alternative mixer skins. A further option for changing the Mixer skins is to click on the System menu in the Mixer window (the Samplitude symbol in the top left corner of the Mixer window). The "Hardware controller (view page 365)" function is also located here.

Mixer keyboard shortcuts

- **Cursor buttons:** Use the cursor buttons on your keyboard to select the various mixer elements and select active elements (fader, knob or switch).
- **Page up/page down:** Changes the value of the active mixer element. Hold the "Ctrl" button simultaneously to change the value in large increments, and hold down the "Shift" button for smaller increments.
- **Home:** Resets the mixer element to its preset (passive) initial value. If you press "Home" again, the element is reset to the last value, providing a simple comparison between the changed and initial preset states.
- **End:** Opens the assigned sub-window of an element, e.g. the EQ window for one of the EQ knobs. This function corresponds to right clicking a knob with the mouse.
- **Enter:** Opens the numeric entry field for a control. Switches are turned using the "Enter" key and "Page up/Page down" keys.
- **Del:** An activated plug-in slot may be reset via the "Del" key.

Mixer operation with mouse and keyboard

Left click: Selects a control element.

Right click: Accesses either a context menu for the corresponding control element or a dialog with additional settings.

Ctrl + left click You can use this shortcut for multi selection.

Shift + left click: You can select all similar control elements between the previously selected and the current control element (multi selection).



Hint: After selecting several mixer elements, you can form these into one group (view page 167). Use the **"Group selected controls"** button to do this.

Ctrl + Shift + left click" on knobs and faders: This keyboard shortcut switches the behavior of corresponding faders or knobs within a linked group (inverse selection).

Some mouse movements create opposing movements of two grouped faders or move panorama faders of two tracks in the opposite direction.

Ctrl & 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 entry box to allow manual entry of numeric values.

Double clicking on control knobs: Resets the control element to its preset (initial) value. Double clicking again sets the value back to the altered value.

Left clicking on value displays: Clicking on the area below the faders or on the left button corners decreases the values. If the mouse is held down, the value is automatically reduced until the key is again released or the end of the scale has been reached. (Shortcut: Page down)



Clicking on the area above the faders or on the right button corners increases the values. If the mouse button is held, the value is automatically increased until the key is again released or the end of the scale has been reached. (Shortcut: Page up)



The level faders can be adjusted very precisely. If you click on these, whilst keeping the left mouse button pressed and moving the mouse pointer left or right from the fader you can fine tune the settings by moving the mouse on a vertical plane. The steps 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 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 like with 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 (shortcut: "Y") under "Keyboard, menu & mouse -> Mouse".

New AUX bus in mixer: 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.

Drag & drop in mixer: You can rearrange individual channel strips in the track number box or name box via drag & drop. The mouse pointer turns into a hand. The overall settings of a channel can be transferred to another channel via dragging & dropping the "FX" button. The EQ 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 effects sequence for VST and DX plug-ins can be changed within the track via drag & drop.

Note: If a certain effect is already available, a repeat drag and drop of the same effect onto the same track performs a reset of the effect to the values of used effect, and does not copy it; the original values will be overwritten.

If all you want to do is move shift effects between the channels in the mixer instead of copying them, simply keep the Shift button pressed while dragging & dropping.

Channels

Each track in the VIP is fed through a specific mixer channel. The sound settings of the respective mixer channel are applied to the objects on the track.

Each channel strip provides the following settings options. Each individual section can be opened and closed by clicking on the corresponding arrow symbol.

In (Input): In this field you determine the recording mode for the incoming signal:

- If you select **"1+2 - Stereo in"**, the channel input will be switched to stereo and the signal will be picked up on two channels.
- If you select **"1 - Mono in"**, the channel input will be switched to mono and the signal will only be picked up on one channel.
- If you select **"Mono mix in"**, the channel input will be switched to mono and the signal will be picked up on two channels and then mixed down to one.
- **"Mono effect processing"** enables you to switch the processing for the complete channel from input to panorama fader to mono. This results in all object effects and track effects being calculated before the panorama fader in mono. This function saves CPU resources, given that it is supported by every plug-in. Opening **"Mono effect processing"** switches the input to mono automatically. This function corresponds with the mono switch in the track editor. This option is not available for buses because they always calculate in stereo.



In: Specifies the audio input for recording.

Gain: The mixer adjust the input amplification for the respective channel.

AUX sends: This section allows you to activate and set up the AUX component for different AUX buses (view page 172). Open a menu by right clicking for advanced settings such as Pre fader send, "Sidechain send" or AUX pan editor". The upper "AUX" button above turns all AUX paths either active and inactive, which effectively makes it an AUX bypass switch.

Inserts: Here you can add effects to the channel strip. Clicking on the arrow right of the accordant Insert Slot, opens the selection menu. The "Insert" button above activates all effects (FX inserts + plug-ins). A visual indicator (*) is displayed for plug-ins that were previously active and are active the next time "Insert" button is pressed.

Equalizer: This section lets you change the sound of the track signal by means of a 4-band parametric equalizer. The knobs change the volume of the filter band.

You can numerically edit the gain and frequency of the corresponding filters.

Right-click on one of the dials for fine-tuning. Samplitude shows the EQ dialog in which you can specify exact settings.

Panorama: Here you can set the direction of the signal in the stereo panorama. The settings of the knob have a different effect on mono and stereo tracks. When set to the center position, the track signal is held in the center of the panorama irrespective of the whether the track is mono or stereo. If the knob is moved away from the center position, the mono signal is moved to the left or right within the stereo field. For a stereo track, the alignment between the left and right channels within the track signal changes.

Beside the panorama dial, you'll also find a switch to reverse the phase.

Right click one of these two control elements opens the stereo panorama dialog (view page 220) to adjust additional settings like panning laws or the stereo width.

Detailed information about the stereo panorama dialog can be found in the chapter "Stereo panorama dialog (view page 220)".

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 this keyboard shortcut **Ctrl + Shift + left click on knobs and faders:** you can switch the behavior of corresponding panorama knobs within a linked group (inverse selection).

Automation: This button gives you access to the channel's automation functions

Detailed information about automation is available in the chapter "Automation (view page 327)".

Solo: The solo button mutes all channels with the exception of the selected channels. Right clicking enables the selection of the output device for the channel.

Shortcut: Alt + S

Solo exclusive: This setting switches the active track to "Solo exclusive" mode, i. e. only the active track is audible. All other tracks are silenced in solo mode.

Shortcut: Shift+ Alt + S

Detailed information on the global solo modes can be found in "Global buttons -> Global solo modes" (view page 171).

REC: This button activates the track. Right click to determine the recording mode for the incoming signal. (see "**In (Input)**").

Mute: The mute button mutes the selected channel. Right clicking enables the selection of the output device for the channel.

Shortcut: Alt + M

Mute/Inactive: Mutes and also deactivates the track. This increases performance as caching and FX processing of this channel are not necessary.

Shortcut: Ctrl + Alt + M

Mute bus inputs: Use this function to mute the bus inputs routed to this channel.

Volume fader: Controls the track levels. The behavior of the volume fader in the MIDI track is preset on controller 7 (MIDI volume). Alternatively, you can also specify that "Volume fader" mode does not change for MIDI tracks or that the volume fader for MIDI tracks

should match the MIDI velocity scaling (view page 244). 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 (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. For this reason, the volume fader may be assigned differently.

Hint: You can also enter levels higher than 6 dB numerically.

Monitoring: Click the loudspeaker symbol to activate the monitoring function. Audio material will be passed through the inputs of the sound card and forwarded to the outputs.

If the "Hybrid Engine" is set up for monitoring, all mixer channel effects for the feeds can be used. This requires working ASIO drivers for the sound card. Samplitude can therefore 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 27).

Track name: You can enter the name of the VIP track here. This name can be edited by double clicking.

FX: This key activates the effects routing dialog for the corresponding track, and can be used to configure and edit the effects chains. This specifies the sequence of the effects.

More information about this is available in the chapter "Buses and routing -> Order of effects calculations and signal manipulation (view page 173)".

Right clicking the "**FX**" button opens a context menu. Here you can open the effects routing dialog for the channel, as well as preset effects settings. Track effects settings can be copied, inserted, reset, saved, or loaded. Save your personal track effect settings in the program directory in "FX presets -> Track FX". Of course, new subfolders may also be created.

Out: Sets the channel's audio output. This may be either a submix bus in the mixer or a playback device.

Master section

The master section can be found to the right beside the channel strips.



Master plug-ins: This function can be used to set effects inserts, MAGIX plug-ins, VST or DirectX plug-ins for the master out signal. Right click on the control button to show the plug-in dialog. To switch the plug-in effects on or off quickly, click the control button.

De-hisser: The "De-hisser" mastering effect helps reduce existing distortions like high-frequency hissing. Open the "De-hisser" dialog by right clicking it.

FFT EQ: A right mouse-click opens the FFT filter dialog.

Vocoder: Here you'll find the Vocoder for the entire signal. A right mouse-click opens the dialog.

Room simulator: The room simulator for the entire signal. A right mouse-click opens the dialog.

Multi-band Dynamics: A right mouse-click opens the entire multi-band dynamics section.

Advanced dynamics/limiter: Right click to open the compressor/limiter dialog for the master signal.

Master Equalizer: This Equalizer affects the sum signal.

Stereo enhancer (SE): The multiband stereo enhancer is used to modify the entire signal's stereo picture. Three bands are available for the bass, mids, and high frequencies.

Mono: This button plays the entire signal in mono. Its purpose is to verify mono compatibility.

Normalize (N): This is the master normalization. If this key is selected, the output level will be adjusted in such a way that the loudest signal sections reach 0 dB levels. The maximum level reached during the last playback will serve as the basis during this process and will be displayed on the peak meter.

Hint: Click the level display in stopped state, and the play cursor will jump to the position indicated by the level.

Link: This key connects the left and right channels of the master signal.

Faders: Both of these faders adjust the left or right master signal. Reset the faders to the 0 dB position by double clicking them.

FX: Opens the effects/routing dialog for editing and configuring any number of effects chains. This specifies the sequence of the effects.

More information about this is available in the chapter "Buses and routing -> Order of effects calculations and signal manipulation (view page 173)".

Mix to file while recording

The **Mix to file** and **On** buttons may be used to mix down in real time 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. "Multi-track 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 further buttons for adjusting the global settings of the Mixer window and Mixer handling can be found at the right side of the Mixer window.

This includes 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.

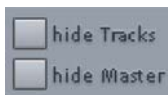
Hold the "Shift" key down and click on one of the "Start track" buttons to save the current view (a left click is sufficient the first time).

The previously saved section may 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 170)



Show functions: Individual sections of the mixer window may be switched on and off to the right of the master volume fader.



Group selected controls: These two buttons are used to group and ungroup elements of the mixer.

To group track controls, select the desired elements while holding down "Ctrl" and then activate "Group track controls". To ungroup, choose one of the elements in the group and then press "Ungroup controls".

Note: If you have previously selected multiple tracks, this is also available to group selected controls in case the same control elements are involved. More about grouping tracks in Samplitude is available in "Working in the project window -> Selecting/grouping multiple tracks (view page 98)".

Multi-channel selection: Multi-channel selection in the mixer functions exactly like multi-track 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 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.



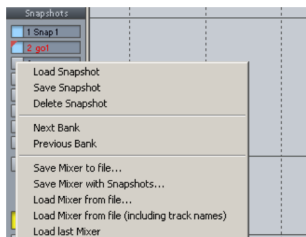
AutoRec: This display will light up if the level automation is written in Master.

Bypass: Using this button you can switch off active effects in the project for comparison purposes.

Solo/Mute: All solo/mute functions can be globally activated and deactivated 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.



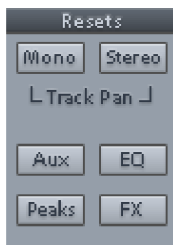
Snapshots: The Snapshot Function is particularly useful to compare various Setups with each other. You can easily switch between setups. Naming snapshots is possible by double clicking the name field. Right clicking a snapshot opens a context menu to **save, load, or delete snapshots** or to **switch snapshot banks**. Using this process, up to 32 mixer snapshots can be saved.

Save mixer to file: Save mixer settings to a separate file via this menu item.

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 may 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 may be recalled via the command "Load last mixer". This makes an A-B comparison between the snapshot and the settings possible.



Reset FX: This enables you to remove effects from the signal path for all channels.

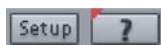
Reset peaks: This enables the LED peak meter to be reset (peak hold display).

Reset EQ: Resets all equalizer settings.

Reset AUX: Resets all AUX portions of the channel strips to their default values (= no AUX portion).

Reset (stereo): Resets all mixer settings to the default settings for stereo tracks (when working with stereo wave projects). This reset function refers to the track panorama settings (view page 220).

Reset (mono): Resets all mixer settings to default settings for mono tracks (when using mono or LR wave projects). This reset function refers to the track panorama settings (view page 220).



Setup: The mixer setup dialog can be opened by clicking the "Mixer setup (view page 468)" button.

?: Click this button to open the help file, which includes detailed information about the mixer.

Solo/monitor volume fader

A solo and a monitor level is available in the master section of the standard mixer.



The solo fader controls the monitor volume in "Solo" mode, while the monitor fader influences the level on the monitor output. The audio monitoring point may also be set using the AFL/PFL switch.

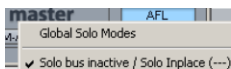
- PFL = Pre Fader Listen
- AFL = After Fader Listen

In general, the monitor bus behaves like "Main to Monitor", meaning that the content of the master may 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 method has been available 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. "Solo" mode is normal during mixing to identify instrument tracks and edit them.

Switch to "Solo in place" mode by deactivating the solo output bus and switching to "Solo bus inactive/Solo in place" in the device selection under the AFL/PFL button.



Caution: When using the monitor fader in this mode, it's possible to influence the audible master level and thereby the level of external devices connected to the stereo master. You can set the monitoring volume in the mixer without using external volume controls. The audible result is different than the peak meter display in the mixer, which is the reason for the numerical level values in the mixer displaying in red. Additionally, the solo fader influences the volume of the solo tracks that are played via the master area. The internal level for bouncing and burning, however, remains unchanged.

Another function of this mode in Samplitude is to toggle between "Pre-fader listen" and "After fader listen" in "Solo in place" mode by clicking the AFL/PFL switch.

Solo/monitor bus (in the "Hybrid Engine" only)

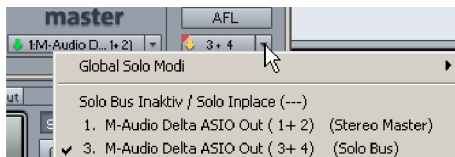
Use this bus to monitor tracks individually by pressing the corresponding solo buttons without influencing the stereo sum signal. This also affects mastering or live mixing.

The signal may be accessed at two different positions:

PFL: PFL (pre-fader listen) accesses the signal before the faders and effects, but after the input gain.

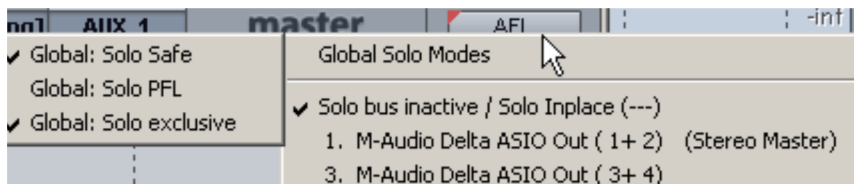
AFL: AFL (after-fader listen) includes all panorama changes/fader movements and influences the monitoring signal accordingly.

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 may 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

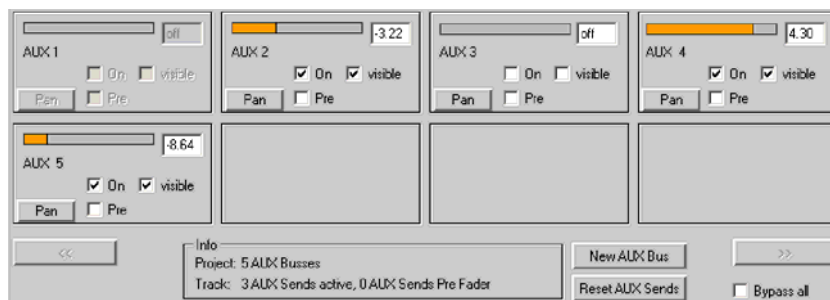
A submix bus combines several tracks. It controls the volume, panorama, and effect settings of all tracks that are "routed" to the submix bus. For example, all drum tracks (hi-hat track, bass drum track, etc.) may be combined as one submix bus so that the entire drum kit may be controlled via the volume controller of the submix bus.

Creating submix/aux buses

- An AUX/a Submix Bus can be created in the Mixer at any time.
- 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 **access 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 feed the signal of the previously created AUX section into the newly created AUX bus.
- To integrate a submix bus, click with the right mouse button on the number of the channel in the mixer. In the context menu, select **"Insert tracks -> New submix bus"**. The submix channel is integrated behind the channel that is accessed. If the output of several previous channels is now routed through this bus, the volume of these signals may be controlled via the newly created bus.
- Right clicking the number of a channel provides an option to select between the properties "AUX bus", "Submix bus", or even both for the respective channel strip.

- Submix or AUX busses are always meant for stereo busses.
- Submix and AUX buses may be used for all channel effects, including volume, panorama automation, and AUX send. The outputs may be routed for any channel to output devices or other submix buses with higher channel numbers.
- Automation of AUX and submix busses works the same as on all other channels.

AUX Routing



By right clicking on a mixer channel's AUX button the AUX routing dialog will open. Right clicking on the channel number will cause a dialog to appear above, if you select "Track effects > AUX sends" in the context menu 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 are routed "post" in Samplitude. Activate the corresponding "pre" box for pre-routing. Use the effects routing dialog (view page 174) 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 220) of a track. This enables the stereo width to be changed or the phase to be inverted for the AUX send, for example.

New AUX bus: Creates a new AUX bus.

Reset AUX Sends: All AUX settings are reset.

The sequence of effects calculations and signal manipulations

The sequence of gain, panorama, and filter manipulations has no effect, at least when working with 32-bit float. In this case, it is not important whether you filter first and the change the gain, or vice versa. Most other effects are also independent on the input signal. This includes the Room Simulator, Stereo Enhancer, Delay, Resampling, and Time Stretching or Pitch Shifting.

All dynamics functions and those that remove or reduce noise (De-hisser and noise reduction) are dependent on the input signal level and, in some cases, on the frequency response of the input signal. Once the optimal settings for a parameter have been found, other effects located in front of the effects should not be adjusted. For example, use the De-hisser and the multi-band dynamics section in the mix master only after all other changes have been applied to the individual objects and the mixer channels.

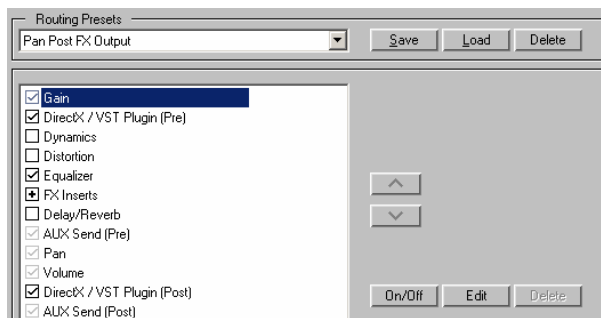
Effect routing/plugin-ins dialog

Open the effects routing/plugin-ins dialog to specify the most important settings for real-time effects/plugin-ins. This dialog is available at object level ("**FX routing**" in the **object editor**), channel/track level ("**FX**" button in the track editor), and at mixer level ("**FX**" button in the stereo channel strip).

Effects Sequence/Plug-ins

All available effects are displayed in the effects list in the respective context (track, object, or master).

The **sequence of the effects** may be configured freely via the effects routing dialog. Open this dialog via the "FX" buttons in the track editor, track box, and mixer.

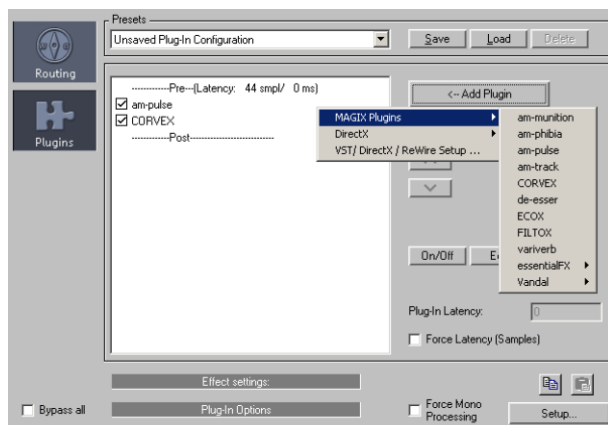


You can set new sequences using the arrow keys. Order the FX inserts before or after the EQ or the dynamics. The free movement of the effects in the insert slots of the track editor or the mixer channel is limited by this order.

Note: Effects inserts may only be moved as a group.

Right click on an effect in the effect routing dialog to access the settings dialog directly.

You can add plug-ins via the "**Plug-Ins**" dialog:



Plug-ins pre or post

The plug-ins dialog enables you to introduce effects "**Pre**" or "**Post**". Highlight "pre" or "post" in the plug-in list before loading a plug-in. The plug-in is then inserted "pre" or "post" fader, respectively. Effects may be moved freely into the track editor or mixer channel's insert slots, but they are limited by the fixed position of the plug-ins in sequence (view page 174), e.g. **DirectX/VST plug-in (pre)** and **DirectX/VST plug-in (post)**.

Effect routing/plugin dialog parameters

Save/load/delete: Manage your individually customized settings here.

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

On/off: Switches of the corresponding activated effect or the activated plug-in. Activating or deactivating the checkbox preceding the corresponding effect/plugin has the same effect.

Edit: Opens the dialog for the active effect/plugin. Another way to open an effect/plugin dialog is to right click the desired effect or plug-in's name.

Detailed information about the plug-in dialog is available under "Software instruments/VST plug-ins/Rewire -> Plug-in panel - graphical display (view page 319)".

Delete: You can also remove internal Samplitude effects (insert effects) and plug-ins instead of just deactivating them, if you want to save resources.

Reset: Returns all settings to the Samplitude basic settings.

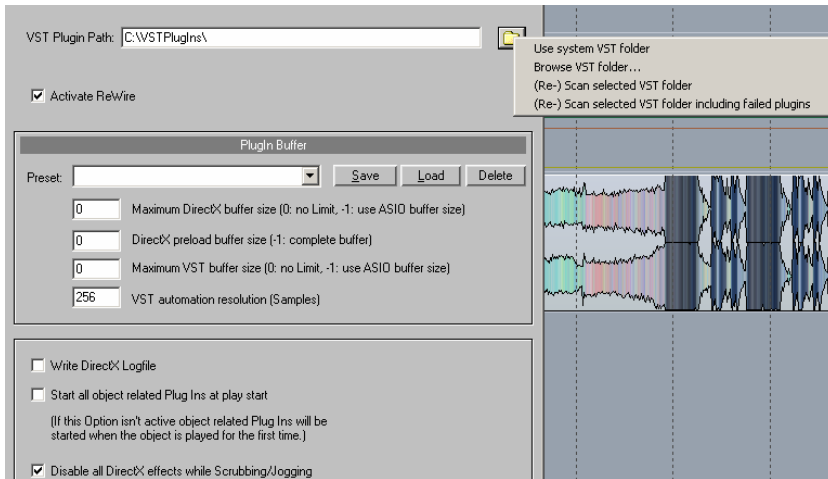
Copy/paste: Use these buttons to quickly apply the settings to different tracks/objects.

Play/stop: Starts and stops playback at the current cursor position.

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; time shifting no longer occurs when plug-ins are used. If the values reported by a plug-in remain incorrect (time delays result from the use of a plug-in in spite of this), then you can force a specific latency value to compensate.

Plug-in options -> Settings: Opens the **VST/DirectX setup dialog** in the plug-in options area.

VST / DirectX / ReWire setup / Plug-in buffer



VST plug-in path: Set the path to your VST plug-in effects and VST instruments here. If you click the folder icon, a context menu will appear. This allows you to **use the VST folder** already set for your **system** as the plug-ins path or select any other **VST folder**. The selected folder is then scanned by Samplitude for VSTs. Plug-ins are then checked for usability within Samplitude. This scan is only necessary once per folder, since available VST plug-ins are saved during this process. If multiple VST plug-in folders are present on your computer, then you may enter additional sources by scanning them with the option "**(Re-) Scan selected VST folder**" once more. If individual plug-ins are not integrated as expected, then try using the option "**(Re-) Scan selected VST folder including failed plug-ins**" to check them. Samplitude detects newly installed plug-ins automatically, but only in the specified folders. The corresponding data is saved in the file "VSTplug-ins.ini".

Activate ReWire: If this option is selected, rewire-compatible client applications can be integrated into Samplitude as synthesizers.

ReWire is generally used for sample-exact, real-time transmission from audio channels between two programs. Both programs can be connected with the same sound card. The transport functions of the applications (such as playback and forward/reverse) are connected via ReWire. If you are working with ASIO drivers, you can direct sounds from ReWire-compatible applications to different sound card outputs.

To integrate ReWire applications into Samplitude, find the ReWire-compatible instrument that you want to connect to Samplitude in the "MIDI out" slot of the track editor via "New instrument -> ReWire". The ReWire application's instrument outputs can be opened via the "Audio in" slot for each track.

The tempo is always oriented according to the settings specified in Samplitude, i.e. Samplitude is the master. The tracks/channels connected to the ReWire-enabled software may be edited with the equalizer, effects, and other plug-ins, and then routed to the available buses.

Note: Please make sure that the sample rates set for the applications connected via ReWire match one another so that playback may be guaranteed at the correct pitch.

Additional information about ReWire is available in the chapter "Software instruments/VST plug-ins/ReWire -> ReWire client applications (view page 325)".

Plug-in buffer

The default presets usually cover most cases when working with plug-ins. Try out various presets if you are experiencing any problems with plug-ins or DSP cards. You can specify the buffer size directly that is sent to the plug-in for XD and VST plug-ins. The following special values apply in this case:

0: The program specifies 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.

DirectX preload buffer size: If you enter the value **"-1"** in this case, the entire buffer size will be used for DirectX start initialization.

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.

The option **"Write DirectX logfile"** is for support purposes. This may normally remain inactive. You may also activate the option **"Start all object-related plug-ins at play start"**. Some plug-ins cause delays when switched on; in this case, these plug-ins should be switched on before starting playback, since otherwise errors may occur when the corresponding objects are played.

Disable all DirectX effects while scrubbing/jogging: DirectX plug-ins may cause errors when scrubbing or jogging. Use this option to disable communication with DirectX plug-ins.

Effects – organization and workflow

In general there is a difference between effect types. The following effects are available in Samplitude:

Realtime effects / Virtual effects / Non-destructive effects

(accessible via track editor plug-ins, object editor plug-ins, the track header plug-in selection field, the "Effects" menu (in which case a check may not be placed next to "Process Effects Offline"), or via the "Insert" section in the mixer)

These effects are calculated during playback or export. The effect settings may be changed at any time to optimize the sound.

Real-time effects differ according to where they are applied:

- Object effects that may be accessed via the object editor or the "Real-time effects" menu
- Track effects/mixer channel effects (able to be accessed via the track editor, track header plug-in selection field, the track header plug-in button, or the insert section of the mixer)
- AUX effects (accessible via the track editor, track header plug-in selection field, or the AUX track header plug-in button)
- Surround effects (able to be accessed via the insert selection field of each Surround bus channel, the track editor, track header plug-in selection field, or the track header plug-in button)
- Master effects (can be accessed via the master section of the mixer)

Internal offline effects/destructive effects

(able to be accessed via the "Effects" menu. A check must be placed next to "Process Effects Offline").

These effects are calculated into the original wave file or a new wave file once. After this time, the settings can no longer be changed.

In contrast to real-time effects, during playback, all resources are used during the playback of the effect.

Here too, the effects can be differentiated according to their location of application:

- Non-destructive audio editing: The effect for the selected area will be written into a temporary file. When saving the wave file, the original and the changed areas will be recombined again.
- Destructive WAV editing: The effect for the selected area will be written directly into the original file. The "Undo" function is available for reversing operations.
- Application in a VIP object: The effect will be calculated for the selected area of the object. Depending on the settings for application of destructive effects, calculation takes place directly in the original material, at the end of the original material, or in a new file.

Plug-in effects

Besides internal effects, additional effects may also be applied as plug-ins.

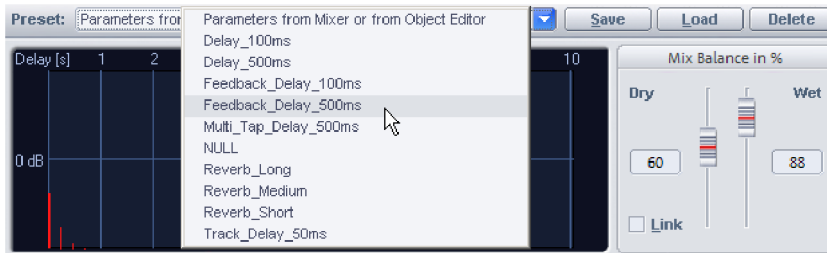
Samplitude supports plug-ins in the following formats:

- MAGIX plug-ins: A selection of included plug-ins, including Analog Modelling Suite (only for Samplitude Pro X) Vintage Effects Suite, Am-munition, essentialFX (efx), Vandal, and VariVerb Pro.
- VST effects (VST FX): This includes all external VST effects
- DirectX effects

These effects can be accessed as real-time effects via the track editor plug-ins, object editor plug-ins, track header plug-in selection field, track header plug-in button, or the mixer's "Insert" section.

Save effects parameters (preset mechanism)

The effect dialogs contain preset selection boxes. These presets are saved and recalled to/from the FX preset directory (a sub-directory of the main program directory) and appear in the selection box after they are stored.



If the desired preset files are not stored in the FX preset folder, these may be accessed via the "Load" function. However, they will not appear in the selection box. Enter the path to the effects preset manually.

Saving new effects parameters

In many new effect dialogs you now have direct access to the automation button and therefore 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 presets.

Here the overview of the other control elements:



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.



A/B comparison function. This control element is only displayed when it is supported by the respective effect.



Reset function. This control element is only displayed when it is supported by the respective effect.



Play button.



Solo button. This control element is only displayed when it is supported by the respective effect.



Here you can call up extra help information.

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.

Effect routing

The sequence of all real-time effects may be adjusted individually for each track. This enables individual sequences to be set for the effects.

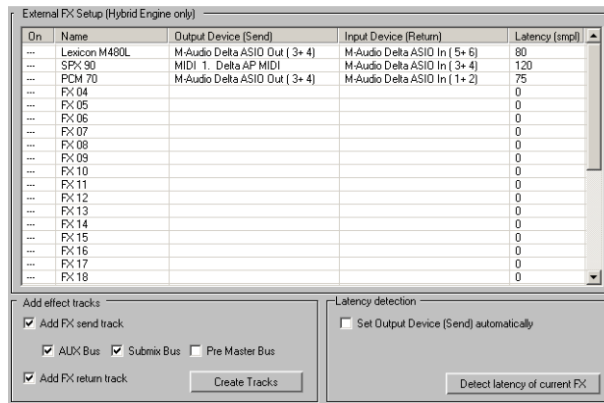
Detailed information about the FX routing dialog is provided in the chapter "Mixer - buses and routing -> Effects routing/plugin-ins dialog (view page 174)".

External hardware effect integration

(for "Mixer FX monitoring/Hybrid Engine" only)

To integrate external effect devices into your setup, you will require a multi-channel audio interface. Reserve a channel pair for every external effect device you wish to integrate.

The integration of external effects or synthesizers can be made via the dialog "File > Program 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 effect latencies so that they can be taken into account during latency compensation.



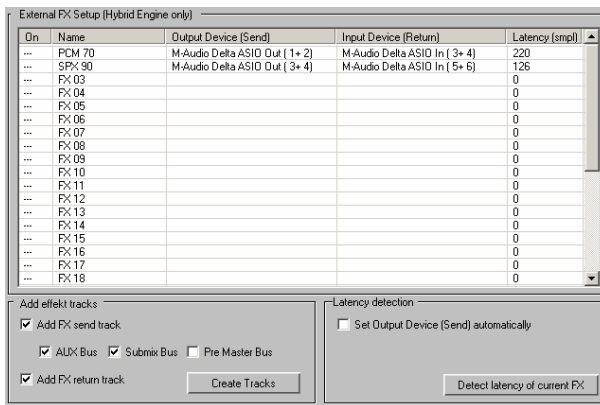
The order of inputs and outputs is saved globally for the program. The setup of the external effect integration is dependent on the project.

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 via the program's own tracks and are not available in the tracks themselves as plug-in inserts.

You can open the "External effects" dialog via "Project options" or via the keyboard shortcut "I".

Routing the effect

First, replace the description FX01, FX02, etc. in the name column with the name of the integrated effect. Clicking the next column opens a selection of the available output devices to feed your effect device as an effect send function. In the "Input device" column, select a free input device as a carrier for the effects return channel. Next, activate the external effect device by clicking the first column and connecting your effect device to the physical audio inputs and outputs that were set up previously.



At all points in the program where input and output channels are set up, you may also integrate external effect devices, e.g. in the mixer, in the track editor or alternately, by right clicking on "Mute" or "Record". In this case, the names of 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

Under "Add effect tracks", select the options "Add FX Send track" and "AUX bus", then tick the box beside "Add FX return track", and finally 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 or return signals of the external effect device.

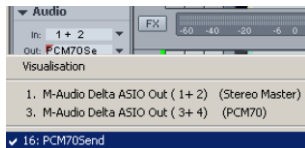


Next, set up the send level of every integrated effect device in the AUX slot for each mixer channel using the mouse, and position the effect signal in the mix using the fader of the effect return channel.

Effect as insert

If you want to add an insert effect, select "Add effects track", then the options "Add FX send track" and "Submix bus", then set a check next to "Add FX return track", and click "Add tracks".

Two new created tracks will be created in the arranger and in the mixer to control the send and/or return signal of the external insert effect device. For the track to which an insert effect should be added, select the send (submix) track for the external effects device as the output device.



Use the effect return channel fader to set the effect signal's return volume.

Note: Make sure input monitoring is switched on for the effect return track. Activate the loudspeaker button in the return track to do this. The loudspeaker symbol is highlighted and you will hear the effect return signal.

The routing settings for each channel may also be adjusted in the track settings dialog. Right click on the track name to start. For the effect send channel, set the output for the sound card connected with the external effects input via "Playback". For the effect return channel, set the input of the sound card that is connected with the external effect output via "Record". For the channel intended to work with the external device, set the effect-send track via "Playback" [e.g. submix: PCM70 Send (track 16)].

Use the double arrow buttons beside the track name field in the track settings dialog to switch the track settings to those of the neighboring track.

It is also possible to use an effect combined as AUX as well as a submix bus. To do so, both options have to be active when creating new tracks. FX sends may be changed later by accessing the track properties (right click on the track header for the FX send bus) and setting a check next to the corresponding setting, i.e. "AUX bus", "Submix bus", or "Pre-master 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 latency of the effect**" 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

Real-time effects at track, object, and master level

Amplitude: Normalize, normalize (quick access), loudness adaptation, fade in/out, set zero

Dynamics: Dynamics, advanced dynamics, multiband dynamics, sMax11, efx_Compressor, eFX_Gate, AM-munition (Samplitude Pro X Suite), AM-track (Samplitude Pro X Suite), AM-phibia (Samplitude Pro X Suite), AM-pulse (Samplitude Pro X Suite)

Frequency/Filter: Parametric EQ, EQ116, FFT filter/spectral analysis, brilliance enhancer (optional), Filtox, De-esser

Delay/reverb: Delay, room simulator, efx_Reverb, efx_StereoDelay, Ecox, Variverb

Time/Pitch: Resampling/Timestretching, Elastic Audio

Distortion: Distortion, Ampsimulation, Vandal, eFX_VocalStrip, eFX_TubeStage

Restoration: De-clipper, de-clicker/de-crackler (optional), de-hisser, de-noiser, get noise sample, remove DC offset (offline), spectral cleaning (offline)

Stereo/phase: Switch channels, multiband stereo enhancer, invert phase (both channels, left channel, right channel)

Modulation/special: Convolution, vocoder, efx_ChorusFlanger, efx_Phaser, eFX_TremoloPan, Corvex, reverse

Sample manipulation: Change sample rate (offline), sample number/2 (offline), sample number *2 (offline), reverse, form loop (offline)

plug-ins

essentialFX: eFX_ChorusFlanger, eFX_Phaser, eFX_Reverb, eFX_StereoDelay, eFX_Compressor, eFX_Gate, eFX_DeEsser, eFX_VocalStrip, eFX_TubeStage, eFX_TremoloPan.

MAGIX Plug-ins: AM-Munition (Samplitude Pro X Suite), AM-Track (Samplitude Pro X Suite), AM-Phibia (Samplitude Pro X Suite), AM-Pulse (Samplitude Pro X Suite), Corvex, Ecox, Filtox, VariVerb, Vandal

Process Only Left Stereo Channel

Process Only Right Stereo Channel

Process Effects Offline: Tick the checkbox here to process effects offline

At object level you can swap object channels, invert phases, Object EQ, Object Dynamics, Pitchshifting/Timestretching, 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."

You can find more information on the Cleaning/Restoration Suite in the PDF document "Cleaning/Restoration Suite" included with the program.

DirectX and VST effects

Microsoft's DirectX and VST-compatible plug-ins may 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

Additionally in the Track Editor the MAGIX synths Revolva2, Robota and Vita are accessible for the first effect slot of each track, plus a list of the included VSTi instruments, object synths Atmos, BeatBox 2, DrumnBass and Livid including connected devices using ReWire. You can include your VST plug-in folder via "Menu File > Program preferences > Effects > VST/DirectX/ReWire (view page 483)" in Samplitude.

Detailed information about plug-ins can be found in "Software/VST plug-ins/ ReWire (view page 312)".

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 **Analog Modelling Suite (AM-Phibia, AM-Pulse, AM-Track)**(only **Samplitude Pro X Suite**), **Vintage Effects Suite (CORVEX, ECOX, FILTOX)**, **essentialFX (eFX_ChorusFlanger, eFX_Phaser, eFX_Reverb, eFX_StereoDelay, eFX_Compressor, eFX_Gate, eFX_DeEsser, eFX_VocalStrip, eFX_TubeStage, eFX_TremoloPan)**, **VariVerb Pro**, **AM-Munition (only Samplitude Pro X Suite)** and **Vandal**.



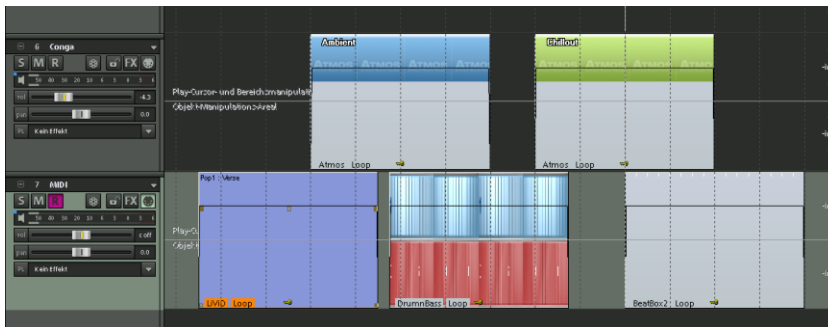
Object synths

Synth objects represent a special form of synthesizer. The instrument selected, i.e. the synth object, is a component of the object created. Objects created in this way are not based on previously created MIDI data. They may be edited easily and can therefore be conveniently used as an audio building block.

Synth objects therefore form individual synth objects. Synth objects may be moved across tracks in Samplitude in any way without changing the synthesizer settings. The synth objects are **Atmos**, **BeatBox 2**, **Loop Designer** and **LiViD**.

Synth objects are available as installation options. After installation, they are located in a separate folder, i.e. "Synths".

Synth objects may be accessed via "**Object -> New synth object**" or via the first plug-in slot of the plug-ins section in the track editor for the respectively selected track. A 4-beat loop object is created at the current play cursor position. The instrument interface for the synth object that is created may be accessed by double clicking the object.



The shortcut "**Ctrl + space bar**" plays the selected synth object in "Solo" mode.

Atmos

Atmos is a synthesizer that can be used to easily create realistic nature sounds. From thunder and lightning to animal sounds and traffic noise, Atmos helps you design natural-sounding atmospheric noises for your projects.



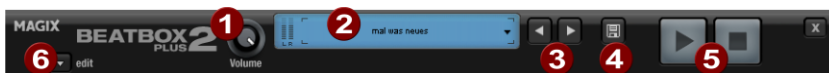
- In the upper edge of the window, select "Scenario", the top category. Select a nature sounds category (e.g. "Thunder and Lightning") here.
- In the middle of the window, a collection of control elements appears for designing the desired ambience. Each element has its own description (e.g. "Thunder") and two faders, i.e. "Volume" and "Intensity". The "Volume" controller adjusts the loudness of the element. The "Intensity" controller adjust the sound, behavior depending on the variable. For "Thunderstorm", for example, this sets the frequency of thunder and lightning; for "Rain", this sets the strength of the downpour.
- The bottom right border features a master volume fader for setting the master volume of the synthesizer. The parameters "Random" and "Auto" are also available. "Random" produces randomly placed settings, and the "Auto" parameter switches on automation of the individual parameters.

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 be 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:

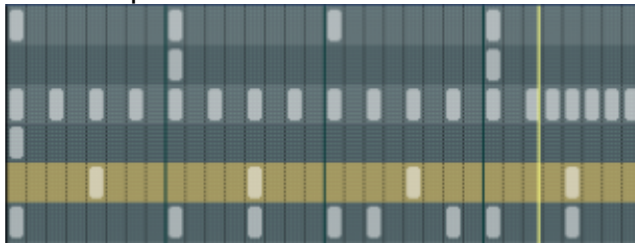
- **Copy/paste:** Copy an instrument from a track and paste it to another one.
- **Empty instrument:** An empty instrument is added. No sound is played, it has no name, and is used to clean up an unused track.
- **Default instrument:** Adds the standard instrument. This features the standard parameter 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. A click on any position in the drum matrix creates or deletes drum notes (events). Clicking and dragging draws in a series of notes. Together with the velocity editing options (see velocity), it's easy to create drum rolls.

If "Shift" is held while notes in the range are clicked, a rectangle will be drawn out which selects the notes contained within this rectangle (lasso selection). Selected notes may 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 selected beat.

Ctrl + shift + double click: Select everything

A single mouse click cancels the selection. The selection is automatically canceled after copying. To keep the selection, hold down "Ctrl" while copying.

Many functions in BeatBox 2 can be controlled with the keyboard; for example, a beat may be triggered with the "Enter" key live in a running pattern. Here's a complete list of the **keyboard commands**:

General:

- e -> Open/close the editor

Pattern editor options:

- a -> "1>2-4" auto copy
- f -> Follow
- 1..4 -> Show beat 1..4
- 0 -> Show all beats
- +/- -> Grid finer/rougher

Selected drum instrument:

- Arrow up/down -> previous/next
- p -> Preview
- Enter -> 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 (see page 249). 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:

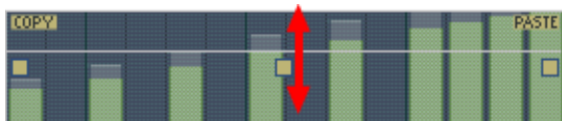
Shift + double click Select everything clicked on in the bar

Shift + Ctrl + double click Select all

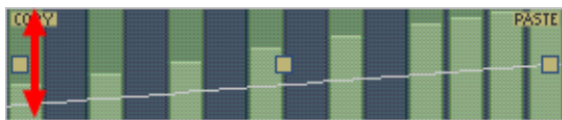


The "Copy" button copies the selection to the clipboard. If you select this or that track in another editor, you can paste the notes there now from the clipboard. If the target selection is larger than the contents of the clipboard, then it will be inserted again. This lets you quickly add a short section throughout the complete length of the pattern.

The three handles allow the velocity or automation values to be edited together.

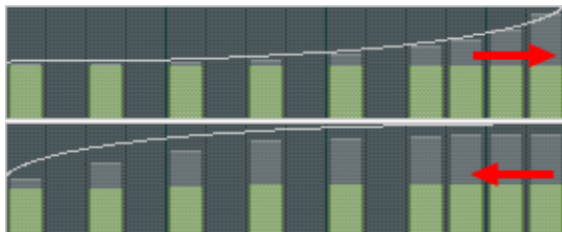


The middle handle increases or lowers the values together.



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 - 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/millisc.)**

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 / millisc.)**

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 synthesizers 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

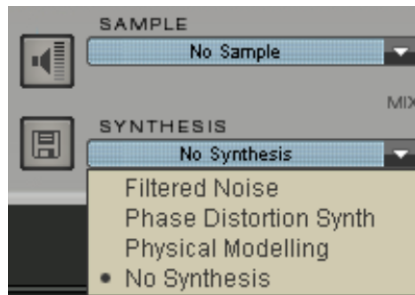
- Input: Input level
- Attack/release: Time constants for compression.
- Ratio: Compression ratio

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 used to create all kinds of drum sounds; the sounds are static and unnatural if the pitch is not altered. This is why the sampler is combined with one of the three synthesis models.



Filtered noise

White noise is filtered by two band-pass filters with separate frequencies and resonance. This algorithm is suitable for creating synthetic snares and high-hat sounds.

Phase distortion synth

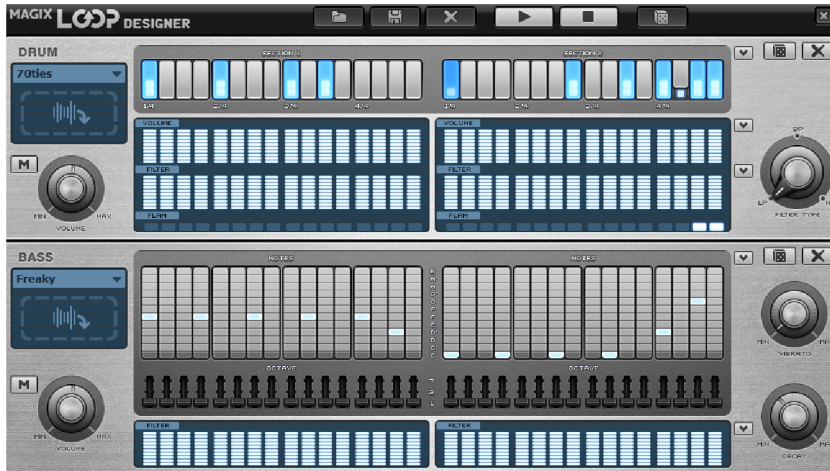
Two oscillators with regulated phase distortion and thoroughly variable frequency modulate each other (FM/cross-modulation/ring-modulation). Depending on the setting, this algorithm may be used to create kick, tom, or metallic percussion sounds; higher values for frequency and modulation level produce noisier sounds for high hats or shaker sounds. Since the oscillator frequency can be set exactly according to the musical pitch, this model may be used to produce bass lines or melodies.

Physical modeling

This is a simple physical model of an "abstract" percussion instrument. A feedback network of delays is caused to oscillate 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.

Loop Designer

The Loop Designer unifies the notable style elements of the Drum'n'Bass genre in a single device: twisted beats and throbbing bass lines. Create authentic drum 'n' bass sounds without previous experience using this machine.



The upper half of the synthesizer controls the rhythm section, and the lower half controls the bass section. Between this on the left edge, a symbol is provided to control both sections or shut them off. For example, the bass section may be switched off so that only the breakbeat in the drum section is added to the arrangement. When mixing the arrangement, only the drum section is added to the mix file.

The right edge includes a "Volume" controller that is used to control the volume of the two sections. The "Play" and "Stop" buttons play your drum & bass creations.

In the "Velocity presets" submenu, help functions are available for programming the velocity sequence (see below for description). Homemade drum loops and bass samples may also be loaded there.

Loop Designer - Rhythm Section

This section makes it easy to create complex and authentic-sounding jungle breakbeats. Jungle breakbeats are produced in professional studios by taking a drum loop and splitting it into little "bites" and then putting them together into another sequence. The Loop Designer makes this laborious process a lot easier.

The top sequence, i.e. the "steps" row, specifies the new sequence. The blue cells indicate the individual sections ("count times") the loop is divided into.

Left clicking one of the blue cells allows one of six icons to be selected. Each symbol indicates other notes or another type of note playback. Each time a blue cell is clicked, the next symbol will be selected.

A short description of the symbols:

- 1: Play drum loop from start
 - 2: Play drum loop from second note
 - 3: Play drum loop from third note
 - 4: Play drum loop from fourth note
- Reverse symbol:** Play in reverse from this position
Stop symbol: Stop playback

Right click the step cells individually to delete them. The "Clear" button the right deletes all step cells; the drum loop is played back in its original condition.

In the "Velocity" row, intensity values of 0 to 16 may be specified with the mouse (left button increases the value, right button reduces it). The manner in which these values influence the loop may be specified via the 3 buttons below the "Velocity" row. If "Volume" is selected, then the velocity value changes the volume in this cell (16= loud, 0 = quiet). If "Filter" is activated, the velocity value specifies the filter strength in this cell (16 = bright, 0 = dull). The "Off" button deactivates the velocity values.

Loop Designer - Bass Section

The bass section produces matching bass lines. Just like in the rhythm section, there are also two step rows here:

- The first row, the "Notes" row, specifies the progress of the notes, i.e. the harmony. Left clicking on a cell opens a popup window for selecting a note. Right clicking deletes a cell. Clicking with the right mouse button on an empty cell produces a "Stop" symbol. This symbol behaves the same as in the rhythm section, i.e. playback of the bass line stops at this position.
- The "Octave" row specifies the octave of the bass line. Octave 1 creates a deeper sound, and octave 3 creates a higher sound. Octave values may only be set, if a note is also present in the row above. Just like in the rhythm section, the "Clear", "Random notes", and a red selection field are available in the upper edge of the bass section. The red selection field may be used to select the bass sound.
- Below the step rows there are two faders for altering the sound:
- The "Vibrato" controller causes the pitch of the bass tone to "sway". If the fader is all the way to the right, the sound will fluctuate more, and if it is all the way to the left, then the pitch won't change at all.
- The "Decay" fader specifies how long it should take until the sound has stopped. Set all the way to the right, the fader causes the sound to dissipate quickly (after approximately 1/4 seconds); all the way to the left, and the sound will reverberate permanently.

LiViD - Little Virtual Drummer

MAGIX LiViD is a great help for creating songs. Only a few details need to be entered for it to play a complete drum track with an intro, verses, refrains, fill-ins, etc., and it provides the exact structure for the entire song. Brilliant stereo drum samples, played by studio pros and adjustable via the "Humanize" effect, create a genuine drum feeling and the perfect sound.



Select one of four styles (pop, rock, funk, latin), which are also divided into four substyles (TYPE), which are further broken down into six song parts per style/substyle (intro, verse, bridge, chorus, fill-in, outro).

Scramble: Creates a random series of four single-beat patterns per style/substyle/song part. "Scramble" regenerates this sequence.

Shuffle: Moves the second and fourth sixteenth of the beat smoothly backwards (100% = triplets).

Humanize: Moves the position of all events randomly forwards/backwards/not at all. Attention! These movements are only very slight and may not be heard clearly!

Snare: Changes the hit style of the snare between "normal", "side stick" (hit on the edge with stick on drum) and "rim shot" (hit on edge and drum simultaneously). Exception: The quietest velocity level of the snare always stays normal ("ghost notes").

Hi-hat: Changes the hit style between "soft" (not opened) and "hard" (half opened). Exception: completely opened hi-hat (pop, type 1, verse).

Ride: Changes the hit style of the ride cymbal between "ride" (on the edge) and "ride bell" (on the cymbal bell).

MAGIX synths

Samplitude includes Robota, Vita, and Revolta 2, software synthesizers based on VST technology

Robota: Eight voice drum computers for "mean" electronic beats.

Vita: A sampler with incredibly realistic-sounding, "classic" 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.

Revolta 2: An analog, varied, and powerful-sounding 12-voice synthesizer featuring a sound matrix, noise generator, and nine effect types. Use this synthesizer to create any electronic music you can imagine.

The following differences define the synth objects:

- MAGIX synths are always loaded to a certain track
- All MIDI objects on this track control this instrument
- All editing options in the MIDI editor are available
- MIDI recordings are possible via the MIDI keyboard

Robota Pro

Robota is a single octave drum computer VST instrument that is based on virtual-analog synthesis. You can access it via the first plug-in slot of any track by selecting "**MAGIX Synth > Robota**".

Samples are not usually used as drum sounds, but rather the sounds are created in real-time with a specialized drum synthesizer. Typical analog sounds of classic drum computers like Roland TR-808, TR-909 or those of more modern manufacturers like Korg Electribe or the Jomox X-Base may be created using Robota. The tool is controlled via a step sequencer with programming or via MIDI.



The Robota drum map can be found in the Samplitude program folder under "MXSynth -> Robota -> ROBOTA.map"

Robota - technical data

- 8-track sequencer
- 64 steps (4 x 16 or 2 x 32)
- "Notes" mode (velocity information per step)
- Snapshot mode (instrument setting can be saved for each step, variable "snapshot recall" time)
- Groove presets ("swing" & "velocity" information)
- "Follow" & "auto copy" mode for simplified editing
- 8 "instruments" (1 per sequencer track). Available for each instrument:
- Oscillator section (sine/triangle/saw tooth/ .WAV) + noise generator
- Modulation (FM/ring)
- Lo-fi section
- Filter section (multimode filter (12/24dB), comb filter, filter envelope)

- Tube simulation
- Compressor
- Volume + pan
- Output routing for stereo output pair 1 4
- Sorting of 8 instruments as "drum kits"
- "Setup" files as overall machine status (pattern + instrument setting + further settings)
- "Config view" with enhanced settings (MIDI trigger, controller matrix, performance parameter, etc.).

Robota - sound synthesis

The eight voices of Robota are setup identically. The synthesis of Robota is quite complicated, since every voice has to generate a large variety of drum sounds, ranging from hissing hi-hats to heavy bass sounds.

The Robota drum synthesizer consists of an oscillator with a selectable wave form (sine, triangular or saw tooth) or a sample. You can also add a noise generator or use this as a pitch modulator (in which case the "noise" controller determines the speed of the modulation).

The oscillator features a pitch envelope and attack/decay controls. It also features frequency and ring modulation. The modulation depth may 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), bit rate reduction (crush) and sample reduction (downsample). The duration of the lo-fi effects may also be adjusted via an envelope (lo-fi dcy). This is followed by a multimode filter (high-cut/band-pass/low-cut) filter featuring a cut-off steepness of either 12 or 24 dB. A comb filter may also be added. You can also modulate the filter frequency via an envelope. A compressor circuit adds "impact" and "power" to the sound ("compressor": intensity; "response": attack and decay) and a tube-amp stage lets you enhance this before it proceeds to the output.



Click "M" to set the instrument to mute; "S" for solo.

You may preview the instrument by pressing the loudspeaker button. Alternatively, use a MIDI keyboard to edit sounds, to trigger the individual voices with "note-on" commands, and to change the individual synthesis parameters via MIDI control change (also explained in "Config view"). Ensure that the "MIDI trigger" button is activated for triggering via MIDI.

(to edit in "Stop" mode, ensure that you have activated input monitoring in Samplitude).

Robota provides 8 audio outputs as 4 stereo pairs. It makes sense to use more than one stereo pair to edit patterns with many tracks and individual sounds separately in Samplitude's mixer. When editing an instrument, choose the output to send to (horizontal slider below the display).

All instruments are routed to the output pair 1 by default. Make sure that the changes are not saved with the instrument but always as a "Setup".

Robota - master section

"Volume" controls the master volume of the Robota. "Tube" adds a controllable tube distortion to give the sound more power and make it sound rougher. The peak meter controls the output level; if this appears in the red area, reduce the master volume.

Robota - sequencer

Classic flashlight programming is used to control the drum patterns, as is the case with practically all classic drum computers and groove boxes. 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. Left clicking switches on the step, and an additional left click deletes the step again.

The maximum length of a drum pattern is four beats. The length may be selected via the small scroll bar above the button list.

Use the appropriate "Edit" button to select the beat you want to edit. Use the "Follow" button to select "Follow" mode, i.e. the step display follows the steps of the currently played beat.

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 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 copy" function, e.g. which makes it easy to add a light variation in the 4th bar.

Programming a new drum pattern:

1. Use the slider to select the pattern length.
2. Select "Event" mode.
3. If you are editing during playback, switch off "Follow". Choose the track you want to edit by using the "Edit" button.
4. Use buttons 1-8 to select the instrument you want to edit.
5. The "Clear" button deletes all tracks in the current bar of the selected instrument.
6. Switch on the appropriate step buttons. Use the "Velocity" slider to define the velocity of a beat.
7. Repeat the process for the other instruments.

Robota - snapshots

It is also possible to automate the 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. Doing this, you may model an individual event within the sound and, for multi-beat sequences, program very lively and “crazy” processes. This enables you to program melodies by changing the tuning of the affected steps (make sure that the oscillators are working in “Chromatic mode” via config view).

Automating a drum instrument using snapshots:

1. Set the editing mode to "Snap".
 - If you are editing during playback, switch off “Follow”. Choose the track you want to edit by pressing the “Edit” button. The snapshot automation should be deactivated beforehand (press “On” button). The settings you have just made would otherwise be replaced by a snapshot already recorded at the current position.
 - 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).
2. Save the sound as a snapshot to one of the step buttons.
3. You may now change the sound of the drum instrument and save the settings to one of the other step buttons.

Note: Parameter changes do not happen abruptly, but are faded internally instead to prevent crackling. Use the "Time" slider to specify how fast the parameters should be faded to the new sound. This allows you to create interesting "morphing" effects. The settings of this controller are saved with the pattern (or setup file).

4. You can jump between the individual snapshots by using the cursor keys if you want to correct individual scenes (input monitoring in Samplitude must be activated).
5. Now click "On" to activate snap automation.

Robota - groove control

The secret to "groovy" beats is to play the individual beats delayed or earlier according to certain patterns. For example, shuffle is used for house beats, in which case the even 1/16s are delayed.

Robota features groove velocity and groove swing presets. Groove velocity presets feature a certain offset for every step of a beat by which the original velocity is increased or reduced; groove swing presets feature a time delay for every step, in order to delay or bring forward the step. The result is a considerably more lively-sounding drum sequence. The strength of each effect can be selected with the % controller.

Groove presets can be selected from the lists in config view.

Robota - setups, drum kits, presets, and patterns

Loading/saving

Instruments, drum kits, patterns, and setups may be selected via the "up/down" keys located next to the display. Alternatively, you may click the display to open a list with available presets. To save a preset, click the "Save" button (disk symbol). The current

preset name will be selected. However, you may also enter a new file name and delete unused presets from the folder view.

(Note that Robota always uses the predefined directories, even if you switch to somewhere else. You cannot save files in a different location.)

Robota organizes sounds and beats according to the following hierarchy:

- An individual drum sound is saved to an “instrument”.
- The 8 instruments may be saved together as a “drum kit”. The individual instrument parameters, however, are not saved, but rather the 8 references to the file names. Therefore, if you manually delete an instrument from your hard disk or change one of these sounds later and save it again, this will affect the drum kits that this instrument uses. “Drum kits” are therefore suitable for organizing frequently used sounds. If you have prepared your own drum sounds by changing existing presets and would like to save these as a drum kit, save the instruments as new presets and then the new drum kit.
- All note information and snapshots are saved in a pattern.
- The combination of everything (instruments + pattern + further settings) produces a “setup”. In this case, it's not important which instruments belong to which drum kits and whether you have made individual changes to the parameters. A setup always saves the current master status, including all sound parameters, notes, snapshots, and groove settings.
- In general, all edit processes may be saved as a “setup”. This would “freeze” everything safely to continue from this position at any time. We recommend the following instead:
- Create a personal “instruments” library.
- Group regularly used instruments as a “drum kit”.
- Create a sufficient basis of “basic patterns”.

This lays the groundwork for future projects. You may now use “setups” specifically by using these primarily the basis for songs. The advantage is that in addition to triggering individual events by MIDI notes, you may also changeover the patterns and drum kits. (Please also read the section about config view).

You can switch individual setups in Samplitude in the preset list (below the instrument selection) or via MIDI program change (1-128).

If you save a project in Samplitude, the last selected setup is opened. Please make sure that you save the state in the setup before closing the VIP.

(When using program changes, note that the storage locations may move when inserting new setups). Setups do not have fixed program locations, but are sorted according to the alphabet.)

Robota - config view

Use the wrench symbol in the top right of Robota to access the back of the device.



Change the following settings here:

- “MIDI play” section:
- Set the MIDI receive channel to trigger the 8 voices externally
- Specify the “trigger keys” for the individual instruments
- MIDI controller matrix: The left list features a selection of „MIDI Control Change (CC)” numbers for operating the controllers of the currently selected instrument externally. By default, this list contains the default CC numbers of the M-Audio Oxygen 8 keyboard. The right list displays the available parameters of the engine for every source. You may customize the controller matrix and even route one source to several targets.

Robota - PATTERN/DRUM KIT PLAY section

- Select the “Play” channel. This does not necessarily have to correspond to the “MIDI play” channel, depending on how the MIDI working environment has been arranged.
- Select the “trigger key” and the corresponding pattern or drum kit. This lets you “play” patterns or kits on the keyboard.
- “Change”: Select the time when the change is to take place (immediately, with the next beat, or always on bar 1).

Robota - GROOVE PRESET section

- “Swing”: If necessary, choose a swing preset from the list. The level of the presets may be adjusted in the sequencer section via the % controller.
- “Velocity”: Select a preset to modulate the volume of your pattern.

Robota - MACHINE GLOBALS section

- Accuracy vs. CPU: “Normal” is selected by default; a compromise of an appropriate timing and medium CPU load. “High” is even a bit “tighter”, particularly in the event of intensive use of the filter modulation and the pitch envelope. You should only use “low” if the processing time is exhausted due to other instruments or effects, since the timing and the envelopes are less prioritized.
- “Oscillators”: “Retrigger” resets the phase of the oscillators for every phase so that wave shape and modulation sound identical for every step. When set to “free”, all oscillators are constantly active, and depending on the application, this may sound more lively and “analog”.
- “Tuning mode”: By default, the “tune” controller locks into place on “tonal” values (“chromatic”), so that you can also program melodies, arpeggios, or bass loops using suitable sounds. “Free” tuning is also possible.
- “Knob mode”: Select whether you would like to move the sliders vertically, horizontally, or prefer circular movements. The latter is the most precise option for fine changes while “hor./vert.” is recommended for rough changes.

Robota - CONFIG MEMORY section

- “Save with current setup”: If this button is activated, all changes made in config view will be saved when the next setup is saved, and are restored when this is opened again.
- “Save as default”: This saves the current configuration as “default”. This is useful for restoring the same conditions in the VIP.
- “Restore default”: Use the “Restore” function to overwrite a “Config” setting created using a loaded setup with your “standard”.

Vita

Vita is provided as an installation option.

The MAGIX Vita synthesizer specializes in realistic playback of "real" instruments by utilizing sampling technology. Short samples of real instruments in different pitches, playing techniques, and volumes are used, combined, and played again at the correct pitch.

The VITA synthesizer is controlled via the MIDI objects.

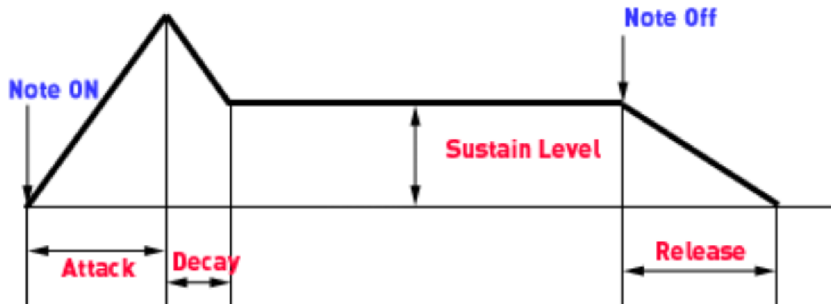
Vita - the interface



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(ecoay) 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.

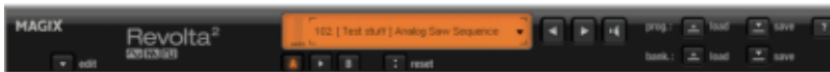
Revolta 2

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).

Plug-ins at track, object, and master level

MAGIX Plug-ins > Analog Modelling Suite: AM-Munition, AM-Phibia, AM-Pulse, AM-Track (Samplitude Pro X 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 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.

AM-phibia is a tube amplifier/channel strip. It combines an optical compressor with a pre and post filter unit. By selecting the filter presets the suitable setting can be chosen depending on the input signal. 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 of the controllers accessible via the "frontplate".

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 via the plug-in section of the mixer, track editor, track, or at the object level via "Effects". Detailed information about MAGIX plug-ins can be found in the help file "Menu effects -> MAGIX plug-ins (view page 698)".

VST/DirectX/ReWire: All DirectX plug-ins installed on your system are written into the Windows registry and are available via the effects slot or the menu item "Effects -> Object DirectX/VST plug-ins". 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 176)", shortcut "Y".

Vandal

(Vandal in Samplitude Pro X Suite / Vandal SE in Samplitude Pro X)

The main concept of the Vandal guitar amp (view page 745) 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

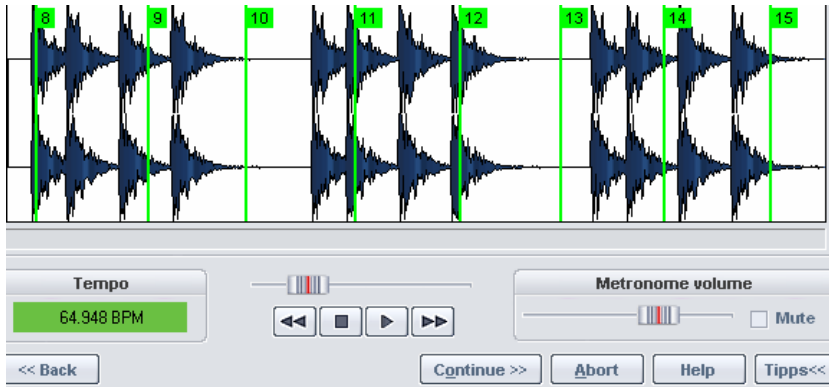
The essentialFX Plug-in Suite (view page 685) offers high-quality audio tools for all-around tasks. Each plug-in has a number of important qualities for producing quick and accurate results:

- Low-resource DSP functions in spite of great 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 via 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_ChorusFlanger, eFX_Phaser, eFX_Reverb, eFX_StereoDelay, eFX_Compressor, eFX_Gate, eFX_DeEsser, eFX_VocalStrip, eFX_TubeStage, eFX_TremoloPan.

Remix Agent

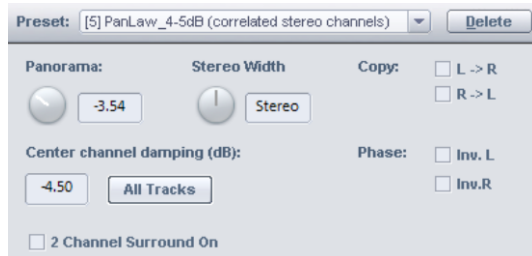
The Remix agent is a powerful tool for determining the BPMs (beats per minute) and the time position of quarter note beats of any song.



Detailed descriptions of the Remix Agent can be found in the menu reference in "Menu > Object > Remix Agent".

Stereo Editor

Right clicking in the panorama controller in the arranger, mixer, and track editor allows you to change panning from the left to right channels for every track in the **stereo editor** as well as further settings related to the phase of the track.



Note: The settings made here are not effective if the track is routed to a Surround bus or Surround master, or if the option "2-channel Surround mode" is activated. In both cases, the Surround editor is used for panning.

For mono tracks, conversion from mono to stereo takes place in the signal flow exactly according to the panorama controls. At this point, effects or plug-ins may be inserted in stereo. The routing positions may be changed in the Effects routing dialog (view page 536).

Stereo dialog options

Panorama: Set level distribution between left and right here.

Stereo width: Adjusts the stereo bandwidth for each track.

If set to the center, no changes occur. If you move the controller to the right, you will only hear the mono portion.

Right click to set the enhancer function for the corresponding track. The middle and side signals will be separated and their volumes edited separately.

Note: Because mono signals have no side signals, this only changes the middle signal with the stereo width control. If set fully to the right, the mono source material will no longer be audible.

Detailed information is available in "Effects -> Stereo/phase -> Multiband stereo enhancer"

Copy L-R

Change the channel assignment of the tracks here. 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 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 value should be selected according to the audio material. Here are some common values:

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. The volume may be perceived slightly louder if the signal is mono.

-6 dB: This setting may be used to automate mono tracks. In the middle, the volume of the right and left channels will be halved.

Common setting for automating stereo signals are **-3dB** or **-4.5 dB**.

The panning law settings can be applied to all tracks by activating "**All tracks**".

Phase: Inverts the phases of the individual channels. However, the switch in the mixer or track editor always switches both channels.

2-channel Surround mode:

Detailed information about this option is available in the chapter "Surround sound -> 2-channel Surround mode (view page 234)".

The individual parameters are processed in the following order:

Copy L-R -> Stereo width -> Phase invert -> Panning/panning law

MS-Processing

"MS" stands for "Middle/Side" and describes the process by which the channels are separated not into left and right, but into middle ("M") and side ("S").

MS recordings

With MS microphone setups files will be received after recording with separate middle and side parts, rather than a left and right channel. With the stereo panorama dialog's presets provided you can edit MS recordings (here we assume that these are available as a stereo file with M left and S right).

MS processing of mono original material

To extract the correct stereo picture from an MS file, duplicate the output material and save it as a new track. For the first track, select the **"Left channel only"** preset. This results in the M signal being played in mono only. On the second track, use the preset **Side signal (Stereo) (from M/S source)** to only play the side signal in stereo, i.e. left +side and right - side. Both tracks can be mixed together each at 0 dB

MS processing of stereo original material

Another requirement is the independent editing of mids and side parts even if the output material is available in stereo format. To do this duplicate the output material and save it as a new track.

"Mono (get mid signal from stereo source)" extracts the mid part from the output material as the first track, while **"Side signal (Stereo) (from stereo source)"** extracts the side part in stereo as the second track. Here you can subsequently mix both tracks each with 0dB.

Advanced MS processingIn 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 (mono) from stereo source: Use only the mono side component of the stereo signal used in the mono track. This way, you can apply all subsequent object and track effects to the side part as mono parts.

Convert side signal (mono) to stereo: Converts the mono side component to stereo.

Surround Sound

Samplitude offers several possibilities for mixing audio material in various multi-channel formats.

One of the pre-requirements is the use of a mixer configured with a Surround Master corresponding to the target format. So-called target formats are 5.0 ITU, 5.1 ITU, or 7.1 SDDS (for example).

In "Project Surround Setup" (File > Project Properties > Mixer Setup > Surround Master > Setup) you can route the channels of the Surround Master to any playback devices you like. The audio material of the tracks in an arrangement may be distributed across the existing Surround buses by right clicking the "Pan" button in the track.

Objects can also be routed directly to the Surround Master and distributed in the Surround Panorama irrespective of the track they are assigned to.

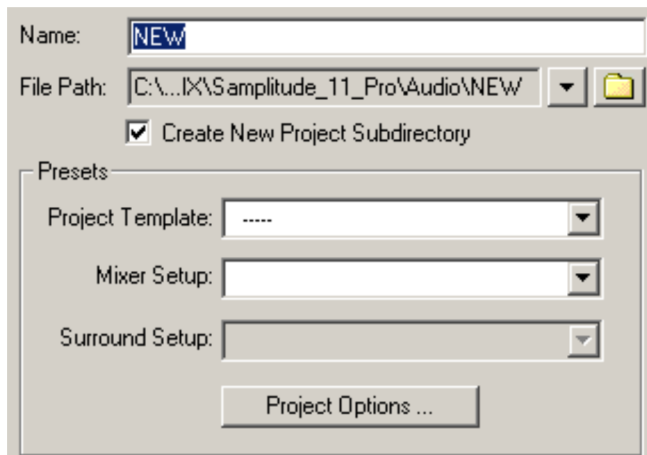
The Surround Panorama module provides four different modes for sorting the signal into the Surround panorama. Various surround effects are provided in the Mixer for mixing in multi-channel format.

Samplitude also offers 2-channel surround, which can be used for stereo projects and is based on "Dolby Surround ProLogic". A regular stereo audio card and Dolby Surround ProLogic compatible receiver/amplifier are sufficient for working with 2-channel surround.

Adding a new project with a Surround mixer

Create a Surround project via "File -> New Virtual Project (VIP)"

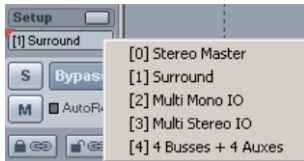
Select "Mixer setup -> Surround" via "Setup for new project (VIP)". Select the desired format under "Surround setup" (e.g. 5.1 ITU).



Besides the audio track's 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 may be displayed by clicking the "Master" button in the mixer. The individual mixer channels may be routed to either the Surround master or the stereo master.

Converting an existing stereo VIP into a VIP with mixer in 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". The "Project Surround setup" opens to select the preferred Surround format (for example 5.1 ITU).

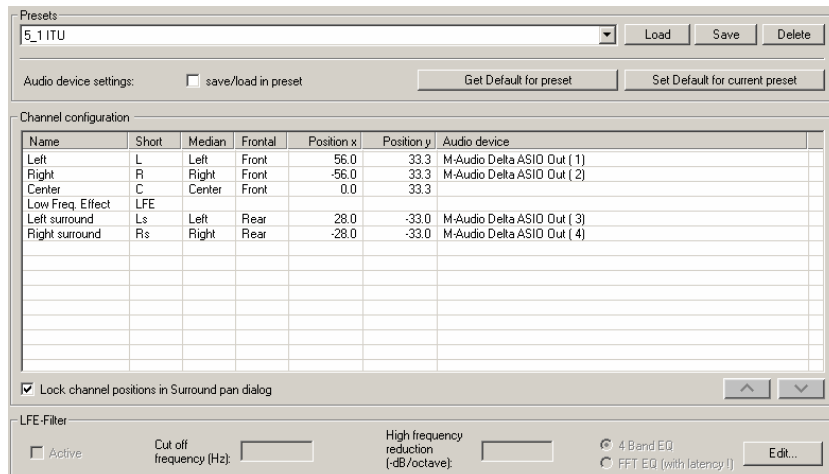


Project Surround setup

Setting the physical output of the surround buses

The "Project Surround setup" dialog opens by clicking the audio device button in the Surround master on the mixer or on the "Setup..." button in the "Surround panorama module" or by selecting the "Setup" button in the master section of the "Project mixer setup" dialog.

The Surround format for the mix is set in the "Project Surround setup" window. You can access various presets (e.g. 5.1, ITU, DD, DTC...) or edit and save your own formats. The filter settings for the LFE 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.

Get default

Pressing the buttons loads the default configuration of the playback devices belonging to the currently active preset.

Set default

The current assignment of the Surround buses to the playback devices is saved as a configuration of the currently active preset by pressing this button. This configuration will be loaded whenever the preset is accessed.

Audio device settings via load/save

Activating this option saves the current playback device configuration during playback, independent of the default configuration in the setup.

Channel configuration

Name: This determines the name of the individual Surround buses. The abbreviation used in the "abbreviation" column is generated from this description. The first letters or subsequent upper case letters of the words are used for the abbreviation.

Left = L
Left Surround = Ls
LFE = LFE

This abbreviation is used in the Surround panorama module and for labeling the corresponding mixer channels. During Surround trackbouncing, the abbreviation is automatically added to file name of the wave files being created.

Example: The file name "Mastermix" is selected when bouncing a 5.1 Surround mix in 6 mono files. The following wave files will be generated: "Mastermix_L.wav", "Mastermix_R.wav", "Mastermix_LFE.wav".

Median/frontal: These columns describe the position of the respective speaker 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. This may be edited by entering the value manually. The settings refer exclusively to the speaker positioning in "Sound field" mode.

Audio device: Assignment of a physical output on an existing sound card for the respective Surround bus. A default device setting may be saved for each Surround preset; see "Get/set default".

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 panorama module (e.g. peak meter), and the Surround effects windows.

LFE settings

You can activate a filter for the LFE channel in "Project Surround setup". Select between a 4-band filter (default setting) and an FFT filter. These filters can be edited by clicking on "Edit".

Cut-off frequency: Determines the cut-off frequency for the low-pass filter.

Dampening: The dampening of the filter above the cut-off frequency is set to 12 dB/octave for the 4-band filter.

Edit: Opens a window to edit the filter settings.

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

Each audio track in the arranger may be assigned to a position in the Surround panorama module via the Surround panorama module of the corresponding channel strip of the mixer. All objects on this track are thereby positioned in Surround panorama at this point.

Object-based Surround panning

Each object may be routed directly to the Surround master or another Surround submix bus via the object editor. The object editor for the current object is opened by double clicking it, and in "Pan/mute/invert", the "Output..." button is set to "Sur master". The corresponding Surround panorama module opens for this to set the desired positioning. Later corrections to the panorama setting may be made at any time via the "Object editor -> Pan/mute/invert -> Surround send -> Edit".

"Stereo" in the "Output" selection indicates routing of the object to the VIP track and the corresponding mixer channel, i.e. track-based Surround panning (basic setting). "Surround master" indicates direct routing of the object to the Surround bus, i.e. object-based Surround panning.

Object-based surround panning is identified in the VIP object display by the "Sur" extension to the object name.

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.

For most applications, an object should be mixed to the Surround bus either via track-based or object-orientated Surround panning. However, by pressing the "Ctrl" key while selecting the output, you can select "Stereo" and "Sur bus" together.

Surround panorama module

The panning of a track or object within the Surround bus is set and edited via the Surround panorama module.

There are several options for opening the Surround panorama module for track-based Surround panning:

- right click the Surround pan display for the corresponding mixer channel strip or
- right click the pan button or fader for the corresponding track in the VIP or
- right click the track information to open the track menu to choose "Pan/Surround editor..." or
- Select the required track in the VIP and choose "Pan/Surround editor..." from the "Track" menu.

The Surround format and the track currently being edited are displayed in the title bar. Example: "Surround panorama module: 5_1 ITU - track 4"

To open the Surround panorama module for object based Surround panning, double click the corresponding object:

Open "Object editor -> Pan/mute/invert -> Surround send -> Edit".

The Surround mode for the object currently being edited is displayed in the title bar of the Surround panorama module.

Display elements

A two-channel input meter is located on the left of the stereo track area in Samplitude. Both scales display the mono signal of the mono input signals.

The level meters of the Surround master's channels are on the right side. Only the levels of the material currently being edited in the Surround panorama module are displayed.

The loudspeaker channels of the Surround setup are displayed as blue dots in the main panorama window. Their position within the Surround panorama field depends on the selected mode. Muted channels are marked with a gray dot. A red dot marks the sound

source that is to be positioned within the Surround panorama. Depending on the "Pan L/R" settings, these may also be displayed as two dots.



A two-channel input meter is located on the left of the stereo track area in Samplitude. Both scales display the mono signal of the mono input signals.

The level meters of the Surround master's channels are on the right side. Only the levels of the material currently being edited in the Surround panorama module are displayed.

The loudspeaker channels of the Surround setup are displayed as blue dots in the main panorama window. Their position within the Surround panorama field depends on the selected mode. Muted channels are marked with a gray dot. A red dot marks the sound source that is to be positioned within the Surround panorama. Depending on the "Pan L/R" settings, these may also be displayed as two dots.

Activating the options below the main panorama dialog display the following information:

Name: The loudspeakers are described using the abbreviations specified in the "Project Surround setup" window.

Volume: Level information is displayed for each loudspeaker. This displays the value of the signal share that is passed on to this bus by the sound source. In stereo modes, the sum of the levels of both sources is always displayed. Holding the "Shift" key and clicking on a source can briefly display the value of a single source only.

Sound field: The meaning of this display depends on the module used by the Surround panorama module.

Channel: Mutes or changes the level of the bus outputs of the Surround panorama module to the Surround buses. Deactivates bus outputs/speakers are displayed as gray dots in the panorama field.

LFE: Level share of the signal that is reproduced via the LFE channel.

LFE only: The source is routed only to the LFE channel.

Center: This parameter controls the portion at the center for assignment of the sound source to the front channels. In certain areas (e.g. film sound), it is normal to reserve center

channel dialogs and mix music and sound in the center. A signal placed directly in the center is played back in 5.1 format only in the center if center = 100%; at 0%, this is only a phantom sound source that is output by the channels L and R. This parameter is often referred to as "divergence".

Pan L/R: Mono and stereo sources may be positioned via the Surround panorama module. The "Pan L/R" function determines how the mono and stereo signals are be positioned. A distinction must be made between the types of signal. For more information, please read the chapter entitled "Stereo and mono signal processing in Surround projects".

Panorama modes in the Surround panorama module

Panning in the surround panorama module can be carried out in different modes. This makes editing of different panning processes easier. Stationary sources of sound can be positioned in the surround panorama, and you can also design the most diverse movements of sound sources.

Note: The different panorama modes work according to different background principles. Changing to a different mode can therefore partly change the acoustic location of the sound source. Exception: The precise level ratios are adopted when changing to matrix mode.

The size of the "Surround panorama module" is scalable. Under certain circumstances some settings can be made more precisely if the screen display is set to a higher resolution.

In all modes (exception: "Matrix" mode) the source of sound (red dot) is moved to the required position in the panorama. The x and y positions can also be changed by entering the value into the corresponding number box, vertically dragging the mouse pointer over the active number field or by using the mouse wheel.

The movements of the sound source can be coordinated and restricted holding down the following keys and moving the mouse pointer:

X + Mouse movement: Only position changes that are parallel to the x-axis are possible.

Result: L -> R movement

Y + Mouse movement or Z + Mouse movement: Only y-axis movements are possible.

Result: Front -> Rear movement

C + Mouse movement: The distance of the sound source to position 0 (x=0, y=0) remains the same. The result is a circuit.

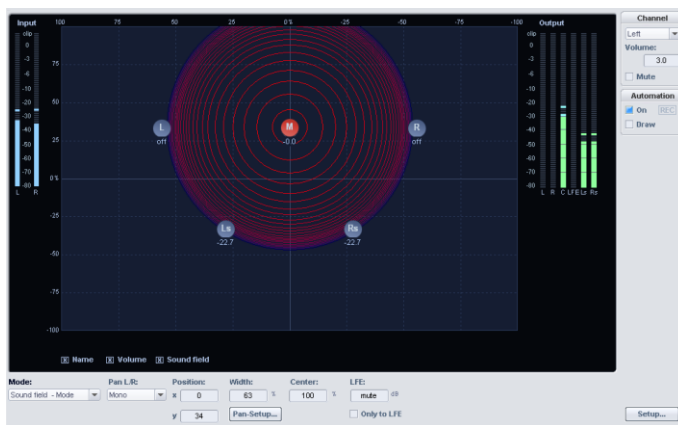
Result: circular movement

A + Mouse movement: The sound source can only be moved with constant angle. 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.

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 so that the distance from a single speaker 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 that cannot be achieved in other modes.



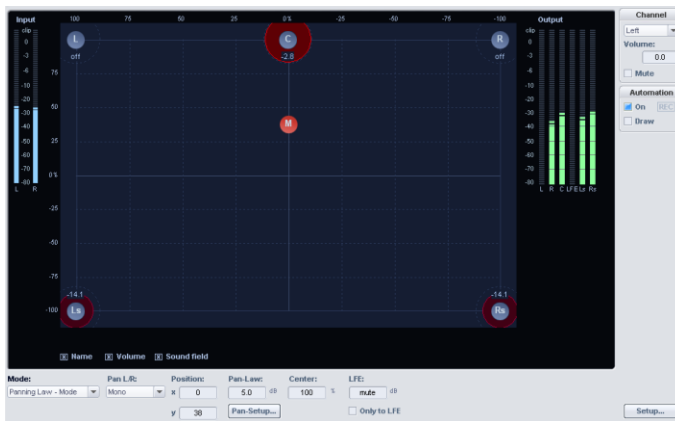
Example of use: Very accurate localization, particularly for movements

Con: Level conflicts, e.g. movements in a constant direction ("Angle" mode is more suitable for this)

Width: Influences the sound field width of the loudspeakers

Sound field characteristic (pan setup): This adjusts the character of the sound field. "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 operates with a display of the Surround panorama as known from many digital mixers.
- The positions of the loudspeakers are displayed at the outer borders of the usable panorama. By clicking the "Sound field" option, you can visually display the graphical levels displayed across the Surround buses. The level distribution between two adjacent loudspeakers is performed according to the -3dB law, which causes both speakers to emit a sound source of -3dB at the exact center location.

Example of use: Simple static 2-dimensional panning, rough localization.

Con: No precise localization, not suitable for dynamic panning (automation).

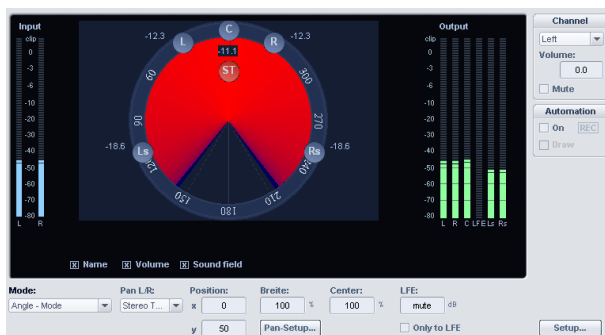
Pan law: The value of -3dB is changed with this setting.

Angle mode

A sound field is displayed which extends from from the center of the circle. The sound source is located on the central axis of this sound field. The loudspeakers are aligned in a circle. The level shares of a sound source that are routed to one channel are determined by the angle ratio between the sound source and the loudspeaker, as well as the aperture angle of the sound field. If the angle of the sound source and the channel correspond with each other (i.e. the middle axis of the sound field is pointing directly at the speaker: angle difference = 0), the level of this channel will be the highest. If the angle difference increases, the level of this channel continuously decreases.

Example of use: Good localization of direction of movements (e.g. airplane passing, etc.)

Con: Not suitable for distance panning.



Width: Determines the size of the sound field aperture angle.

Sound field character (pan setup): You can adjust the character of the sound field. "Inverse logarithmic" creates a larger aperture angle and shorter transitions between the speakers. "Logarithmic" produces a smaller aperture angle and longer transitions between the speakers.

Maximum sum output level constant (pan setup): If this function is active, the total level of all channel outputs of the Surround panorama module will not exceed this value. If this setting is activated in "Angle" mode, level drops in the sound field connected with larger aperture angles will be avoided.

Matrix mode

The level of the input signal sent to the individual surround buses may be entered directly in "Matrix" mode. Entering the values manually by double clicking the numeric field.

Dragging the mouse over the level bars while holding the left mouse button also changes the value. Fine tuning is possible by additionally holding the "Shift" key down.

Example 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.

Con: No automation possible.



Unlike other modes, the "Pan L/R" settings in "Matrix" mode mean:

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 reflection sources will result.

2-channel Surround mode

Two-channel Surround is a method for adding additional room features to a normal stereo signal. No additional audio channels are required. The result is also a stereo format including additional Surround information ("encoded").

Using a Dolby ProLogic-compatible amplifier system (also referred to as Dolby Surround), additional Surround information may be decoded and played back via 4-5 speakers. In addition to the stereo speakers, there are then one or two additional (rear) Surround speakers and an additional center speaker.

Dolby Surround ProLogic only supports one rear channel. If two rear speakers are present, the same signal will be played back on both speakers.

The signal remains fully downward compatible with stereo playback; it may be played on any system not featuring Dolby Surround.

From a technical viewpoint, the output signal is a regular stereo signal. No additional output devices are required and, consequently, nor are Surround buses/Surround masters.

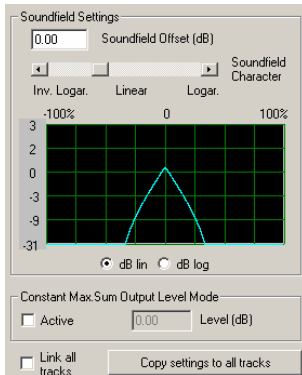
Samplitude is able to work simultaneously with 2-channel Surround tracks and stereo/mono tracks. The left and right signals of conventional tracks are played back in the Surround mix via the L and R speakers, the center channel is fed the mono signal (R+L). For each track, the 2-channel Surround may be activated via the track info dialog or as a parameter in the track settings.

The Surround panning module may be opened by right clicking the pan controller in the mixer window or by right clicking the "Pan" button in the VIP. You may also click the Surround control box in the mixer window (only visible if "Surround" mode is active).

Now select the position of the sound source within the Surround space by placing the red circle at the desired position with the mouse. In "2-channel Surround" mode, the automation functions available in the other Surround panorama modes are also available.

Pan setup

The "Pan setup" window allows you to make further changes to the Surround panorama module. Open this window by clicking "Pan setup" in the Surround panorama module or by right clicking 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 may be compensated if they are too strong.

Sound field characteristic (only available in the Surround panorama module in "Sound field" or "Angle" mode): This parameter is described in the section covering "Sound field" and "Angle" modes.

Maximum sum output level constant (pan setup): If this function is selected, the maximum level of all channel outputs of the Surround panorama will not exceed the set value. This function is especially useful in case of automation and if sources are intended to move throughout the room. Unintentional fluctuations of the max level may therefore be compensated as positions change.

Link all tracks: Changes in the following settings within the Surround panorama module are always transferred to all tracks of the VIP: 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.

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

Frequently used settings may also be saved as presets in the Surround panorama module. The pan L/R mode and settings are also saved besides the position of the sound source.

Use "Load preset" to load previously saved presets.

Stereo and mono signal processing in surround projects

Mono (pan L/R settings): The following applies to stereo signals: a mono sum will be created using the left and right signals. This is positioned as a single mono sound source in the panorama.

The following applies to mono signals: this signal is positioned as a single mono sound source in the panorama.

X-sym (setting pan L/R): The following applies to stereo signals: the left and right channels are symmetrically aligned to the x-axis. For example, this allows you to pan a stereo signal to front L/Ls.

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/R.

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/Surround R.

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 along 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 may be changed by holding down "Ctrl".

The following applies to mono signals: the original and mirrored sound sources are kept at a constant distance from each other and are moved in parallel. The distance between the two sound sources may be changed by holding "Ctrl".

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.

The sum of the levels of both sources is always displayed under each loudspeaker in any mirror mode (if display options are active). Holding the "Shift" key and clicking a source briefly displays the value of the individual sources.

Alternating between stereo and Surround formats

Each channel strip of the mixer may be routed either to the Surround master or to the stereo master. Simultaneous routing is not possible. However, you can use mixer snapshots to mix stereo and Surround more or less at the same time.

For example, first set a Surround mix. This may be saved in the mixer as a snapshot. All panorama settings, filters, etc. are saved.

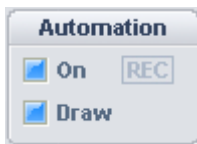
The "Stereo" preset is then selected in the mixer. All mixer channels are thereby routed to the stereo master. If the stereo master fades out in the mixer, it is faded back in after clicking the "Master" button again. All required settings for the stereo mixer may now be set again. This is then also saved as a snapshot.

You may now alternate between the stereo and Surround mixes by switching between the two snapshots.

Both mixtures may be saved with the "Trackbouncing" function as a new wave files (*.wav). Trackbouncing is done separately for each mix, and the respective mixer snapshot (stereo or Surround) must be accessed for this.

Automating the Surround panorama module

Panning movements on tracks set to Surround panning may be automated. To do this, first activate the automation in the Surround panorama module.



Pressing the pan button for the corresponding track in the VIP window can also do this. If the sound source is now moved during playback, this movement is recorded and displayed as a curve in the VIP. This curve may now be edited in the VIP using "Panorama mouse" mode.

Drawing Surround automation curves

To use this automation mode, you must first select the range in the VIP where you would like to add the change. Draw mode is now activated in the Surround panorama module. You may now draw any movement for the sound source in the Surround panorama. This sequence is then applied in the previously selected time span when playing back the VIP the next time. This curve may be edited after in the VIP by using the mouse in "Curve edit" mode.

Effects in Surround projects

Difference between stereo and Surround effects

The following Surround effects are available for Surround buses:

- Dynamics
- Multiband dynamics
- Advanced dynamics
- Room simulator
- FFT filter
- De-hisser
- Vocoder

Effects may only process up to 6 channels in parallel instead of only two (stereo) when working with Surround.

You may define the number of channels by grouping.

Accessing Surround effects

Surround effects may be used in Surround buses (including the Surround master) and in Surround AUX buses.

Grouping mechanism for Surround FX, Surround control group

Grouping and ungrouping: Hold the "Ctrl" key while clicking the channels in the output's mixer for the channels routed to "Sur mast" that you want grouped together or removed from a group.

Add effects to Surround bus (Surround FX): If Surround channels have been grouped, all of the channels in the group will be processed using a single instance of the effect, i.e. the parameters affect all channels in the group equally.

Surround control group The Surround channel buttons are located in the Surround control group that appears with the Surround-capable effects below the effects dialog.

Grouping effects:

- The parameters entered in the effects window affect all channels of the active group (as they would affect two stereo channels in "Stereo" mode).

- If presets are loaded into the presets window, the parameters affect all Surround channels of the group activated in the window but not the Surround channels of other groups.
- For dynamic effects (including the multiband dynamics module), the control signals are generated using all channels of the group (just as they would be generated from the two stereo channels in "Stereo" mode).

Editing Surround sound automation

Automation recording in the Surround sound dialog

When recording an automation, the source signal may be moved to any position in a room. Samplitude records this movement during playback or recording and creates automated events on the corresponding curve in the VIP track.

To change the start position of a reverb source, start the recording process of the automation by clicking and holding the desired position of the reverb source in the Surround panorama module right after playback starts. The source signal position will be automatically registered and the resulting movements will be recorded by Samplitude. Samplitude indicates that recording is in progress with a red "Record" indicator in the window.

Draw mode in the Surround panning dialog

In order to prepare to drawing and move Surround sound as the source signal, a range in the VIP track where the automation will take place must be selected first. "Draw" mode must also be activated in the Surround panning window. Now you can move the source signal to a new position in the panning window. Samplitude creates automation events automatically to generate this movement for the automation curve.

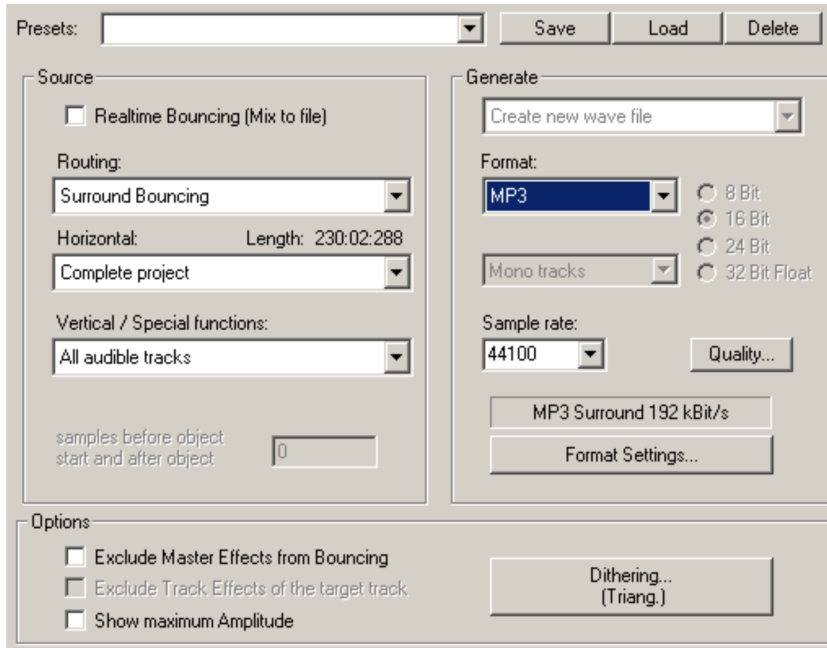
Drawing automation curves in the VIP track

Samplitude allows you to draw new Surround sound automation curves in a VIP track or alter existing ones. Ensure that the "Pan" of the respective VIP track is activated. The "Automation on" feature may also be selected in the Surround panning window. "Panorama draw" mode must then be selected in the tools menu. Use this to draw new curve events in the VIP track. Position the mouse pointer close to an existing curve and click the left mouse button. While the key is held down, record new events for this curve.

Surround bouncing

Surround track bouncing allows you to automatically record complex sound mixes to a new wave file. The outputs of the individual surround buses are used as a recording signal.

The dialog "Trackbouncing (internal mixdown)" opens via "Tools".



Surround Bouncing provides selection between creating a wave file or an MP3 files.

An **interleaved file** may also be created:



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.

The remaining settings may be selected during regular trackbouncing (view page 456). The abbreviations for individual Surround buses set in the "Surround audio devices" will be added at the end of the selected names of files to be recorded.

Example: The file name "Mastermix" is selected when bouncing a 5.1 Surround mix in 6 mono files. The following wave files are generated: "Mastermix_L.wav", "Mastermix_R.wav", "Mastermix_LFE.wav". The following combinations are output as stereo files: "Mastermix_LR.wav"; "Mastermix_LsRs.wav"; "Mastermix_CLFE.wav", and "Mastermix_CLFE.wav".

MIDI in Samplitude

MIDI setup

All global MIDI settings are provided in the system dialog (shortcut: "Y") under "System options -> MIDI".

Please read "System settings -> MIDI settings" (view page 31) for more.

Import, record, edit

Recording MIDI tracks

For MIDI recording and MIDI recording modes, please read "Samplitude Quickstart -> MIDI recording (view page 88)".

Importing MIDI data

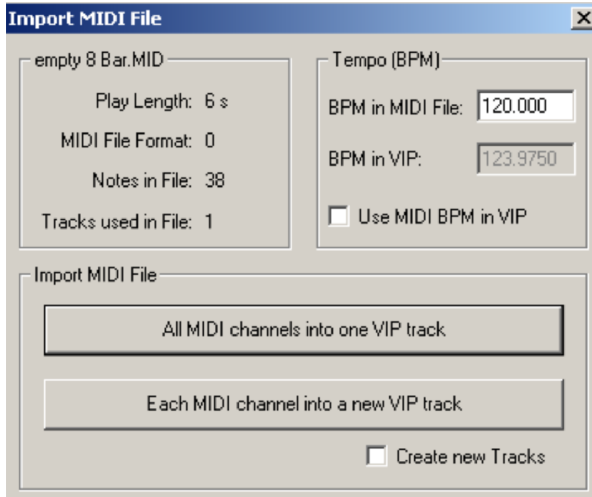
Existing MIDI files may be imported into a Samplitude VIP project as objects.

Menu:	File -> Import -> Load MIDI file...
Shortcut:	Shift + M
Mouse:	Drag & drop from file browser or Windows Explorer

Set the playback cursor to the position at which the MIDI file should be inserted.

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 check next to **"Use MIDI-BPM in VIP"** to do so.

If the option **"All MIDI channels in one VIP track"** is selected, Samplitude inserts the MIDI object into the currently selected track in 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 **"All MIDI channels in a new VIP track"**.

Samplitude also offers the option **"Create new track"** for **multi-track 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.

Freeze MIDI object ("Object > Freeze objects (view page 563)") causes **each selected object to be rendered as a new wave file**. The original MIDI objects are replaced in the arranger by audio objects.

Note: Make sure to check that the your MIDI receiver's audio return signal (VST of ReWire instrument) is routed to the MIDI track.

Freezing is advantageous if a very system-intensive real-time plug-in is activated. The original object always remains preserved and may be re-edited using the "**Edit object freeze**" function or put back into the arranger via "**Unfreeze objects**".

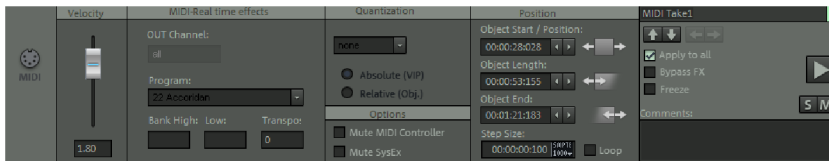
MIDI objects can also be edited using the MIDI object editor as well as in the different MIDI editors such as the Matrix editor, the drum editor, the controller editor, the event list and the note editor.

MIDI object editor

Object: Object editor
Shortcut: 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's possible to control a multi-output VST instrument using a single object on several MIDI channels. Notes in the corresponding MIDI editor can be edited simultaneously or only in one selected channel within this object.



If there is only one channel in the MIDI object, settings will only be available for this MIDI channel.

You can also use the MIDI object editor to quantize MIDI files into MIDI objects.

Note: These settings influence the MIDI object in real time and not destructively, making the changes invisible in the MIDI matrix or drum editor.

Important functions and settings in the MIDI object editor:

Velocity: The velocity fader works comparably to the volume fader in the wave object editor.

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 volume of the notes in the MIDI object to their physical maximum or minimal value (0 or 127), calculating it together with the actual velocity note value. Changing the fader is identical to changing the top-center handle of the MIDI object in the VIP track.

MIDI real time effects: Here you'll find the settings for a MIDI channel:

MIDI OUT channel: Here you can redirect the MIDI files of a channel to another channel.

Program : The values preset here will always be sent if the object is played back again. Use these settings if various objects in a MIDI track should use different programs of a sound generator.

Bank High/Low: This sets the MIDI Bank Change bytes.

Transp.: This button transports all MIDI notes of the corresponding channel.

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 control via the SysEx files.

Position: Here you can specify the **object start / position, object length, object end** and **step size**. 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 tick the checkbox here, then you will set the MIDI object to "Loop" mode. Now drag the bottom left object handle to the left 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 this may also be edited.

In the square field to the side you can select the object color.

Using the **arrow keys** you can "jump" up or down to the object of a neighboring track.

Use the **double arrow buttons** to jump to the previous or next object. 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 an 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 (Shortcut: space bar).

Lock: The activated Lock button corresponds to the key button's switch in the 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 settings** (Shortcut "Y") under "Program ->Object lock definitions" (view page 549). Locking can be temporarily cancelled by pressing the "alt" key.

Solo: This button moves the play cursor, during playback, to the start of the selected object and plays that only. Track-dependent buses and AUX paths are also observed at playback.

Mute: This button mutes the selected object. You can also mute the left and right channels individually too.

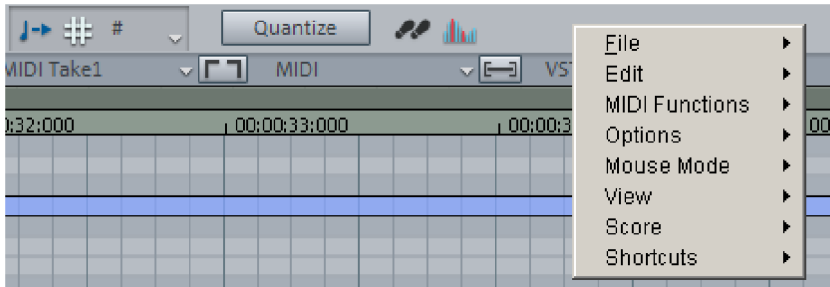
Commentary: You can type comments relevant to the selected object in this field.

MIDI Editors

Open 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.

In case you open the MIDI Editor in the Docking bay (view page 50) there is the option use the toolbar by right clicking in a free range.

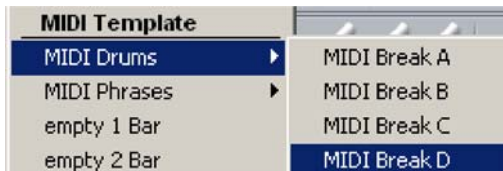


Once opened it displays the MIDI files starting from the currently active position over the length of the object or within the selected range.

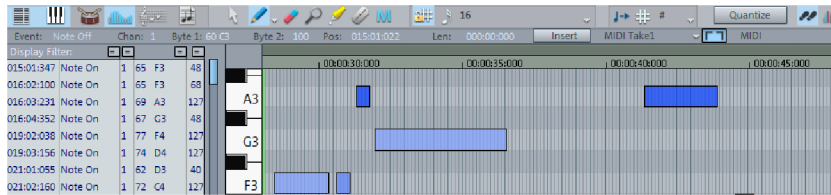
The MIDI editor always shows the current MIDI object and the current arranger track from which this object originates. In the field to the right you should see the MIDI out or the plug-in activated for this track.



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 pre-defined MIDI templates from within the Samplitude "Templates" sub-folder.



Now you can draw in events into the MIDI editor (view page 250) or record notes via your MIDI keyboard.



Detailed information about MIDI recording is provided in the chapter "SampleRate Quickstart -> MIDI recording (view page 88)".

Working with the MIDI Editor

MIDI data can be edited in the MIDI Editor in five sub areas:

piano roll editor



drum editor



Controller Editor (for example, velocity, MIDI volume...)



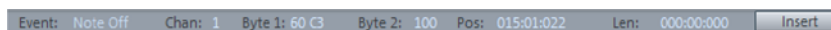
Event list



Note editor



Various tools are available for editing, e. g. the pencil or eraser tools. Exact values for each MIDI event can also be set in the edit fields via the event window.



Fundamentally, changes such as moving or deleting notes, always refer to all selected MIDI events (red) with just a few exceptions. Changes to the selection in a range always apply to every other range as well. You can, for example, select a group of notes in the Matrix Editor

and then change the velocity of these note groups which modifies all selected notes simultaneously in the Controller Editor.

MIDI editor: Select events, delete events

(Matrix-Editor, Drum-Editor, Controller-Editor, Event list, Note editor)

- Select event: Left click on the event or pull a frame with the left mouse button over the event
- Add/Remove event to/from selection: "Ctrl" + left click on event or "Ctrl" + pull the frame over the new event
- Change/Set current event within multiple selection: Left-click on selected event
- Set current event, deselect all other events: Double-click on event
- Selection of events within an area: Left click on the first event, then left click with held-down Shift key on the last event. Or: Drag a frame over the desired event(s).
- 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 note drawing and lengthening (back and front): Alt
- Select previous/next note: Left/Right arrow keys
- Delete events: Right mouse click

MIDI editor: Copy/duplicate events

You can copy selected events using "Ctrl + C" and then paste them at any play cursor 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 play cursor.

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 488)".

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 via the MIDI editor's "Mouse modes " menu and via the following buttons:

Selection (Shortcut: 1)



Lasso: Hold down the mouse button to draw a selection window.

Click in an empty space to remove an existing selection.

Draw notes (Shortcut: 2)



Draw Pencil

Draw an event by clicking the selected MIDI editor and dragging to the right with the mouse. The event snaps according to the current snap-quantization values. If the "Alt" key is held down, then the event may be drawn freely (without the snap function).

Draw drums (Shortcut: 3)



Drum Pencil

A sequence of notes may 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.

Draw pattern (Shortcut: 4)

Select the MIDI events to use for the pattern first, then save this pattern via the keyboard shortcut "Ctrl + P". Next, draw in a sequence of previously selected patterns with the "Draw pattern" tool. Moving the mouse back (to the left) while holding down the mouse button removes previously drawn notes.

Change velocity (Shortcut: 5)

When this tool active, drag the mouse vertically, and the velocity values of the selected events will be increased or decreased relatively 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 velocity of events by their levels, switch on "**Velocity**" for a controller slot button in the Controller Editor Range.

Eraser/Delete (Shortcut: 6)

This deletes the MIDI notes by left clicking. If multiple notes are selected beforehand, then all notes will be deleted. The eraser tool can also be used on specific MIDI notes to delete them. The eraser is also available to for all other tools (except the magnifying glass) by right clicking.

Magnifying glass (Shortcut: 7)

Zoom in with the left mouse button to the length display horizontally; use the right mouse button to zoom back out.

The combination of **left mouse button + dragging** zooms in on the range. The zoom mode may be temporarily accessed again by holding the "Z" key at any time. After releasing the key, the mouse mode is active once more.

Glue notes (Shortcut: 8)

If a MIDI note is clicked on using this tool, then the note will join with the next MIDI note of the same pitch.

Separate draw notes (Shortcut: 9)

This tool separates a note on the matrix into two notes. The separated note snaps to the next snap point.

Mute (Shortcut: M)



This mode mutes individual notes or selected groups of notes by clicking/activating them again. This function is also available as a command in the MIDI editor menu, "MIDI functions" (Shortcut: "Ctrl + M").

Tips:

- "Shift to pencil": The "Shift" key is now the hotkey for "Draw notes" 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 via the MIDI editor menu **Shortcuts -> Edit shortcuts**.
- "Delete" mode may 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).
- No matter what the present mode is, "Draw" mode may be switched on temporarily by holding down the "Shift" key.
- Notes created in "Draw" mode contain the MIDI channel and velocity specified by the editing fields below 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 always refers to all selected notes. If no notes have been selected, all functions are applied to all notes.

Legato: Notes may be prolonged and played back in bundles.

Shortcut: Ctrl + L

Note Quantize (standard): This command conducts standard quantization of the lengths of the MIDI notes of all selected MIDI objects according to the MIDI quantize settings.

Shortcut: Ctrl + Q

Extended Quantize:

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: This command applies quantization to the start and length 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.

Q length/quantize MIDI length: This command applies quantization to the lengths 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.

Reset Quantization: Use this command to reverse all completed quantization steps.

Hint: "Reset quantization" is still possible after saving and loading a VIP. During quantization, the distance to the next grid position saved together with the note and recalculated after "Reset quantization" is selected. This way the original groove or phase of a note can be restored even after displacement or copying.

Quantization settings: Make detailed changes to the quantization settings in this dialog (see below).

Quantize/Thin out controller: Controller values may be quantized or reduced with this function.

Humanize: This command considers the current **Humanize** value in the quantization options.

Mute notes (mute): Mutes and unmutes notes.

Remove note overlaps (polyphonic): Notes may be shortened to avoid any overlaps. Chords (simultaneously played notes) are recognized and left uncorrected, i. e. chords are not split up.

Remove double notes (monophonic overlaps): Notes may be shortened to avoid any overlaps. This function forces monophonic voice leading.

Transform sustain (CC64) to note lengths: You can transform sustain values of events into notes lengths and display them in the MIDI editor.

Transpose: Here you can transpose the pitch of selected events into half steps.

MIDI Timestretching: With this function you can make various changes to selected events, e.g

- double the tempo
- halve the tempo
- scale to range length.
- manual stretch. This allows you to enter individual values in the **Stretch Factor** field.

If, when using this command you don't select any notes it is the same as selecting all notes.

Retrograde (Reverse): With this function you can reverse the playback sequence. The events move by a further quantization step each time they are reversed again.

This command affects all events inside the specified range i.e. all selected events. If you have not selected anything, all events will be affected.

Melody Inversion: This function mirrors the selected events in terms of the pitch of the current note.

If you don't define any particular notes, the mirroring will affect the mid pitch of all notes.

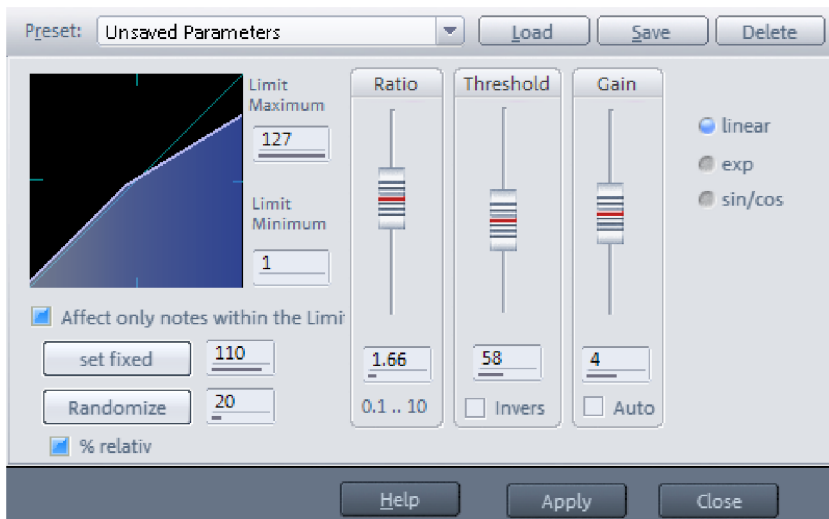
Velocity dynamics

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.

Quantize

Irregularities when recording patterns maybe 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 550)".

Recorded MIDI events may be set to the exact note start and note length values using MIDI quantization. Quantization may 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 in the length quantization values input field you enter the value # the value will be linked to the respective snap quantization value.



The setting **Musical measure** means that, signatures based on quarter notes serve as a quarter snap unit, and time signatures based on eighth notes serve as a 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 259). If no notes are selected, then all notes will be quantized.

Quantize

Shortcut: Ctrl + Q

Right clicking this button opens the dialog for the global grid/quantize options (view page 259).

The command "**Extended 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.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.

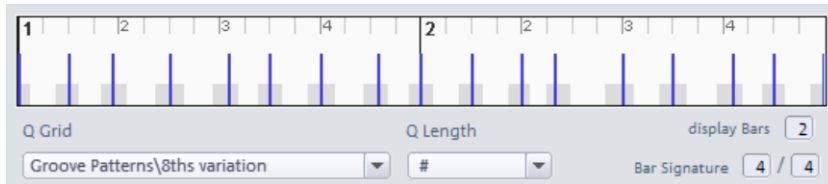
"**Soft Q (approximate quantization)**" includes the current soft Q values in the quantization options.

Quantize Note Ends to Grid This option may 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.

Quantization - window view

In this case, 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.

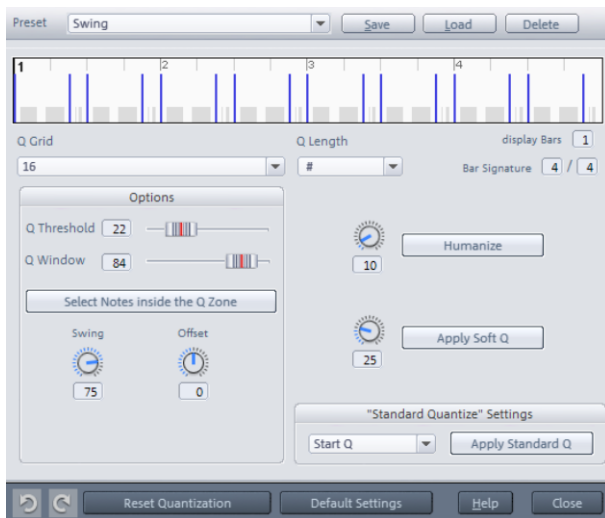


The beat names in the quantization grid depend on the beat signature that is set. The blue lines depict the effective Q snap as well the velocity level (bar height) for groove templates.

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 58)" 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)

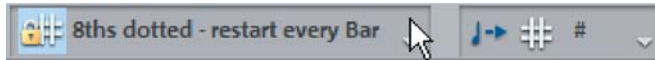
The Q grid also provides n-tuplet quantization values (for thirds, fifths, and sevenths). For example, if **7** is entered, the grid will be adjusted for septuplets. The count will be divided into 7 subsections for quantization. A thirds snap is indicated by a "T" behind the number.

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 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 $\frac{1}{4}$ of the grid width left and right of the grid point ($\frac{1}{16}$ 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" triplets. 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

The "**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 via keyboard or controller keyboard

In the MIDI editor, so-called "Step recording" may also be executed via the computer keyboard or a MIDI fingerboard. 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 via your keyboard. Note length and step length may be specified via 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	: Octave entry up/down
CDEFGAH	: Note entry in current octave
Shift	: Enter chords

Different notes may be entered without the play cursor jumping if you hold down the "Shift" key. This enables chords to be played.

Of course, notes may also be entered in "Step recording" mode via the controller's 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 269)), since these often appear as short, explosive events. Display is limited to the most important information, note starting points, and velocity.

Velocity Mode

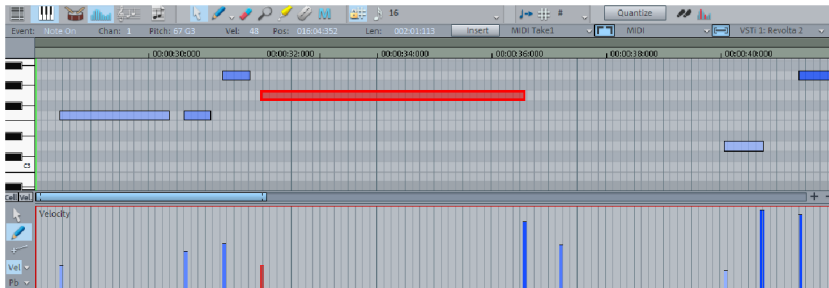


If you click the "Vel" button next to the "Cell" button, the view switches to "Velocity" mode.

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.



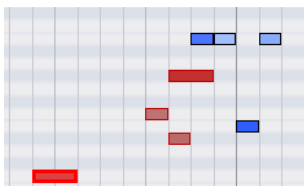
The Matrix Editor is easy to recognize via the piano icon on the left side of the editor. The MIDI events displayed are arranged according to their respective pitch.

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.

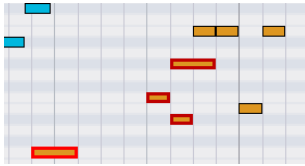
Selected notes are displayed in red. Here too, a more intense color symbolizes increased velocity

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.



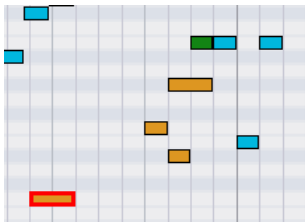
Velocity colors: Unselected note (blue), selected notes (red) and current events (red with border) in the Matrix Editor

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.



Track color display for MIDI events

In the MIDI channel color display MIDI events are displayed in different colors, based on the MIDI channel settings.



MIDI channel color display

Display of filtered events: 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 5. By selecting both MIDI channels in the MIDI editor menu "**Options -> MIDI channel filter**", you may make all notes in channels 2 and 5 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 277) provides additional display filters that only function within the list.

Display of muted events: MIDI object settings (view page 244) (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.

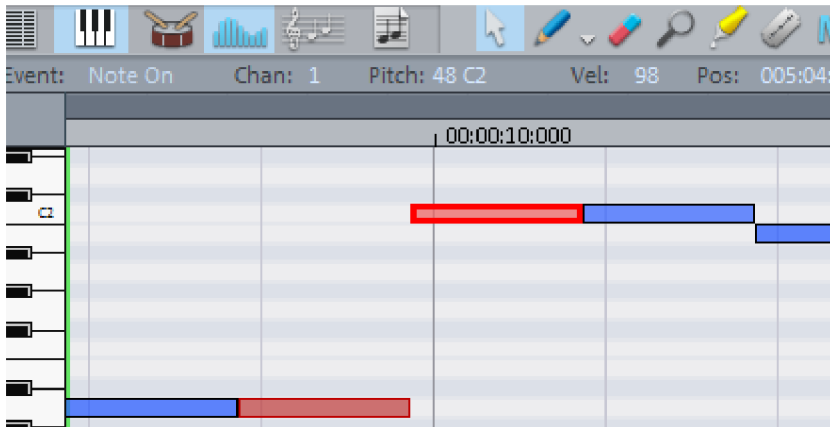
Note: If you additionally press "Ctrl" during any of these selection methods, the new selection is added to the current one (i. e. the previous selection is not canceled).

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.

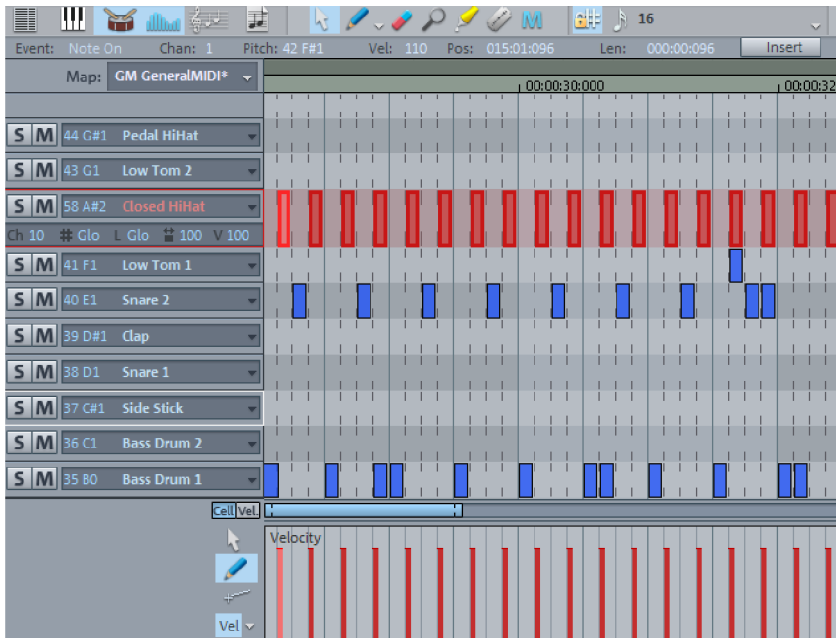


Drum Editor

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 58)), 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 264) activates by default. Here, moreover you can turn Velocity Mode (view page 265) 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 (3): This mode also includes a draw function. However, the quantization length is also included while the events are drawn.

Pattern tool (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 (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 (6): This tool deletes events with a single mouse click.

Zoom mode/magnifying glass (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 certain qualities to the different drum instruments of a MIDI drum kit. Besides key assignment for pitch, the export note, the MIDI channel, and the quantization are defined.

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 under in the track editor under "MIDI".

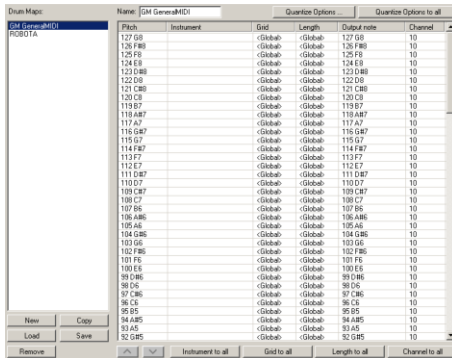


It may be the case that your synthesizer, VSTi, or the drum computer do not refer to the GM map, but to a different mapping setup. If this is the case, you will hear a different sound than you expect when playing the drum events (e.g. a tom drum instead of a bass drum). It is recommended to create a correct drum map for the current playback device. In this case, select the command "Create new drum map) and assign the individual instruments to the corresponding pitch or key assignments of your MIDI keyboard. The quantization options and MIDI channel options may be defined individually. The newly created drum map must be saved, and will then appear in the selection menu.

Note: A drum map is already available for the MAGIX Synth ROBOTA PRO. If you embed ROBOTA into your setup, please select "ROBOTA PRO" as your drum map.

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.

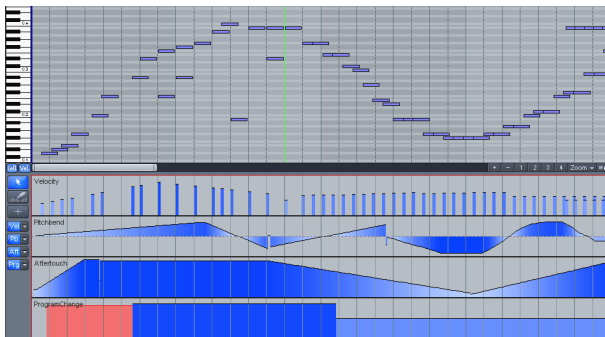
Controller Editor

The controller editor is located below the matrix or the drum editor; this may be hidden or shown via the keyboard shortcut "Alt + V".



ein- und ausgeblendet werden. If you position the mouse below the horizontal scroll bars, the mouse cursor will turn into a vertical double arrow. Pull it 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.



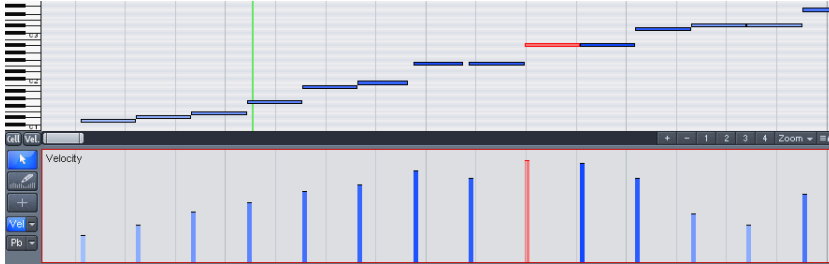
Note: To display more than one Controller button, hold the mouse just below the scroll bar and drag the double arrow that appears upwards whilst holding the left mouse button. If you now click on one of the Controller buttons, the corresponding Controller line will be refreshed.

If you click on the arrow for a controller slot, the following controller types are available for selection:

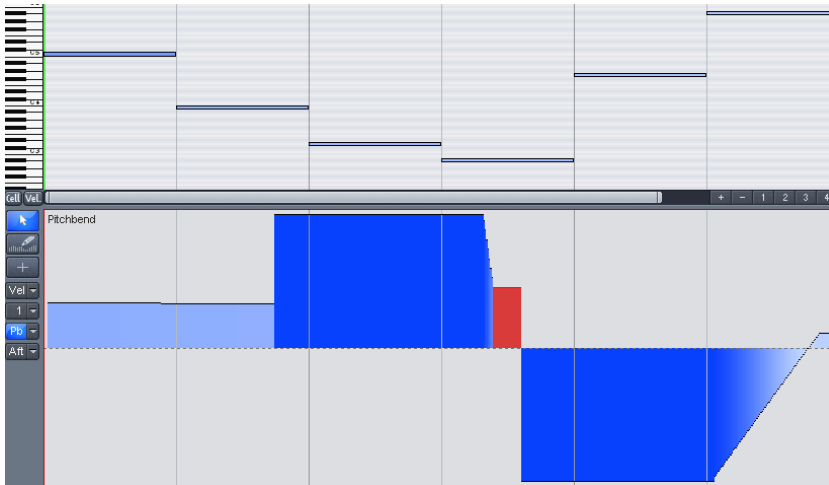


- Velocity
- Pitch bend (bend the note pitch, just like with the pitch bend wheel of a keyboard)
- Aftertouch (pressure on the key after the first strike to control additional parameters, specified via the MIDI playback device)
- Program change (selection of program changes for assigning preset sounds 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 are also represented in red.



Selecting and drawing events

Combi tool:



Keyboard shortcut: Ctrl + 1

The mouse cursor button provides you with a combi tool. Use it to select, change values, and draw freehand and uniform lines.

Click a position on the controller editor and start drawing horizontally, and two red vertical lines will appear. The first indicates the starting point of the selection and the second displays the end point. All events located within the limits are now selected and displayed in red.

Individual events may also be selected by clicking the respective bar.

To **delete** selected **events** in the controller editor, press the "Del" key or the back space key.

Other editing options with the combi tool:

Click on the bar end: The controller value may now be adjusted by dragging vertically. Hold the "Shift" key down while dragging, and all of the selected events will be set to the same value.

Alt + drag (with the pencil): This activates freehand drawing, and the mouse cursor becomes the pencil. Detailed envelopes, ramps, and grades may be created by dragging in the respective controller range.

Shift + drag (with the cross hairs): This activates line drawing and the mouse pointer becomes a cross hairs. 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 may be created by dragging in the respective controller range. Correct curves during drawing by dragging backwards. Any existing (multi) selection is ignored. Create crescendi or decrescendi using velocity curves, for example.

Single clicking creates a new ramp which reaches up to the next controller event.

Shift + click & dragging with the cross hairs to draw a line.

Hint: When editing the velocity, no new notes are generated; existing velocity values are only modified by dragging and clicking the events.

Drawing lines:



Shortcut key: Ctrl + 3

This activates line drawing; the mouse pointer becomes a cross hairs. Create envelopes, ramps, and grades by dragging horizontally.

Single clicking creates a new ramp which reaches up to the next controller event.

Hint: When editing the velocity, no new notes are generated; existing velocity values are only modified by dragging and clicking the events.

Tips

Copying controller events:

You may copy selected events using "Ctrl + C" and then paste them at any play cursor position in the corresponding controller editor using "Ctrl + V".

Timeshifting several selected controller events:

First, select all events you would like to move. To select different controller types together, hold down "Ctrl" at the same time. Now you can move the selection in the time field by entering a value or by vertically dragging the double arrow.

Use the display filter function for velocity values:

In multi-voiced areas, the velocity bars are located above each other, making it difficult to select the respective note bars. 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. 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 "Ctrl" or "Shift" for the keyboard area.

Another option to selectively edit velocity bars located on top of each other is to edit the currently selected note using the mouse. Click on an event inside the piano roll or the controller editor. Finally, navigate to the desired note with the "Cursor left/cursor right" buttons and change the controller value by clicking the top third of the red area and dragging the double arrow that appears.

Quantize controller events

MIDI controller events may be quantized and thinned out; select the "MIDI functions" menu and the "Quantize/thin out controller" command to do this. Quantization occurs according to the quantization settings.

Detailed information about quantization is provided in "MIDI editors -> Quantization".

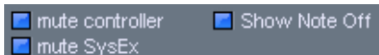
Event list

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.

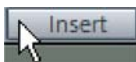
Working with the list editor: The list editor does not only display note events, but also percussion instruments (in drum editor mode), MIDI controllers, and SysEx messages. These controllers and messages may be hidden and even filtered by checking the appropriate “mute” box.



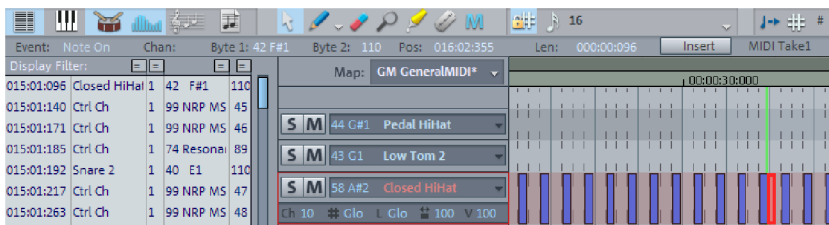
Double clicking on a SysEx entry in the list opens a simple editor for viewing and editing the SysEx information.

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 play cursor position using the “Insert” button.



The values of the previously selected notes are still in effect for the edit fields channel, byte1 (Pitch), byte2 (velocity), time and length.



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:

Event	Ctrl Ch	Ch.:	1	Byte1:
004:03:000	Note On	1	57	A2 110
004:03:096	Note On	1	57	A2 110
004:03:192	Note On	1	60	C3 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:

Event	Ctrl Ch	Ch.:	1	Byte1:
004:03:288	Note On	1	64	E3 110
004:04:000	Note On	1	64	E3 110

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 notes with a velocity of 40 or lower:

Event	Note On	Ch.:	1	Byte1:
004:03:000	Note On	1	57	A2 110
004:03:096	Note On	1	57	A2 110
004:03:096	Note On	1	58	A#2 110
004:03:192	Note On	1	60	C3 110
004:03:288	Note On	1	64	E3 110

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 50).



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 should see the activated VST plug-in for this track.



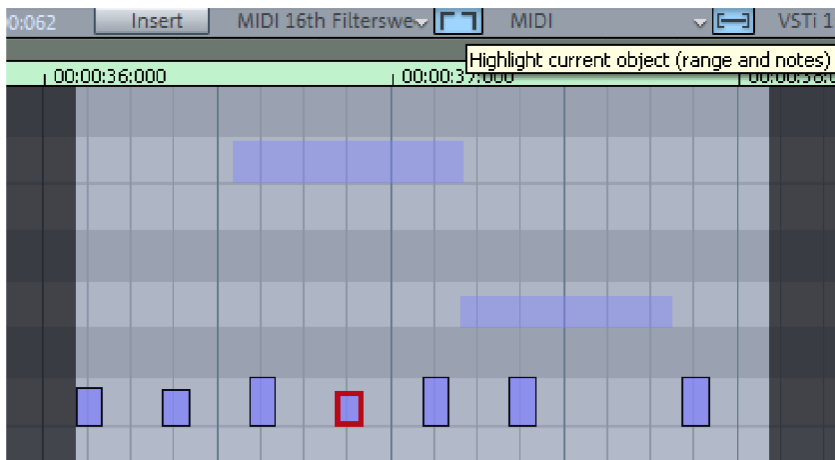
New MIDI events will always be inserted into the current object (i.e. the currently displayed object). In Score mode the current note line is observed.

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.

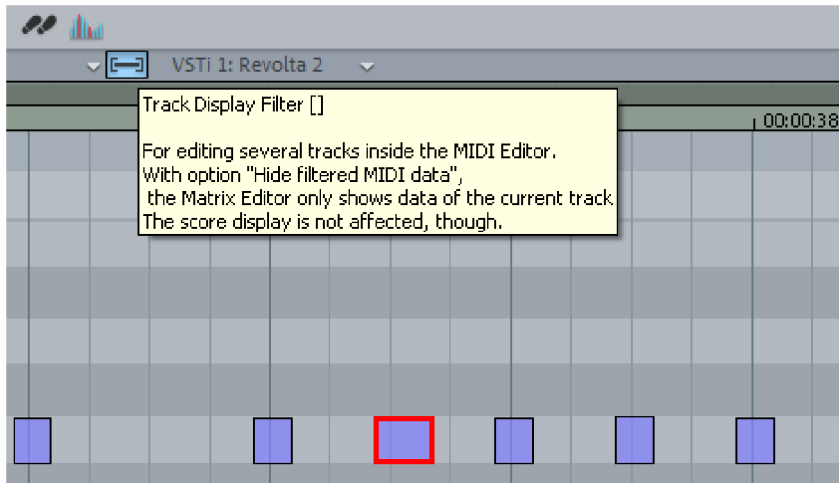
When selecting notes or MIDI events or when clicking using the pen tool the associated object becomes the active object. The current Note line will be automatically converted in the Note editor.

You can also copy and insert MIDI notes between multiple objects.

The **Current object area** can be highlighted using the button behind the object selection field. Areas lying outside the current object will be grayed-out and the notes of other objects will be paler, but still selectable.



A **display filter for the current track** may be activated while in MO editing mode. Data from other tracks are then grayed out. If the option "Hide filtered MIDI data" in the "Options" menu is activated, then MIDI events that are not current will be made invisible.



Note: If the note editor is activated at the same time, only the velocity values of the current staff will be displayed in the controller editor.

In representing notes, the note system of the current track will be highlighted using blue note lines.

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.

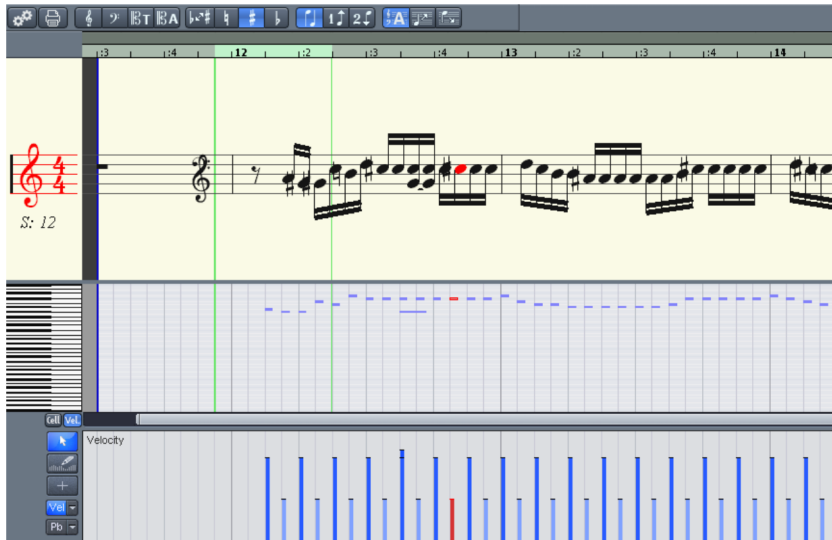
Note editor modes - Linear view



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.

Please 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 staff 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



After the selection of the "Piano" preset

Further explanations of the notation system (view page 291) 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 score are indicated in the matrix by a diagonal line. This function is practical for making trills easier to manipulate, or in order 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

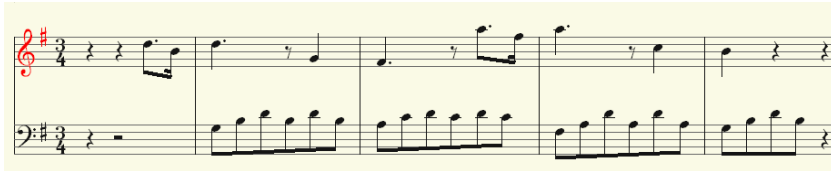
Samplitude automatically generates a notation display from the MIDI events contained in the MIDI object. This is always correct with regard to pitch and position. However, this does not mean that the notation can be optimally read, since displaying note lengths also plays an important role in this context. 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 290)).

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

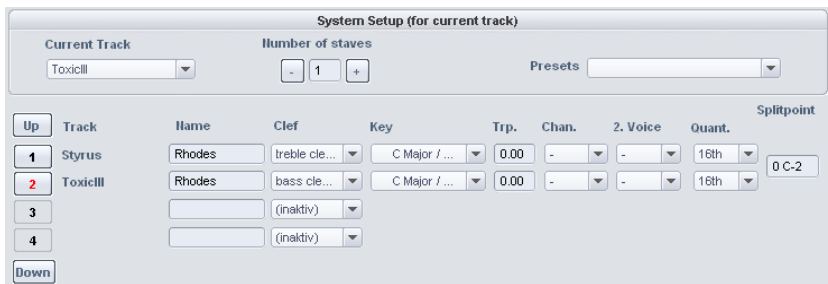
to ungroup them.

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

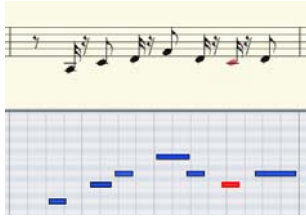
Options (current staff)

<input type="checkbox"/> simplify note lengths (interpretation) <input type="checkbox"/> hide note overlaps <input type="checkbox"/> automatic staccato articulations <input type="checkbox"/> recognize triplets	<input type="checkbox"/> strict measure beat/pulse division <input type="checkbox"/> larger beam groups <input type="checkbox"/> recognize grace notes <input type="checkbox"/> force polyphonic notation
--	--

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.



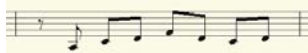
Display with option "simplify note lengths (interpretation)".

hide note overlaps

With this option you can get rid of overlapping of sequential notes which, for example, arises as a result of playing legato:



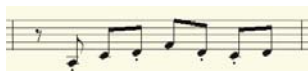
Original



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). Longer notes are displayed as multiple slurred notes as the case may be. This can simplify the legibility of the score.

larger beam groups

Beam 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 "Automatically recognize grace notes" option

force polyphonic notation

The display of the notation always appears multi-voiced.

Notation symbol

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 play cursor 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 301)". 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 format 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.

The screenshot shows the 'Page settings' dialog box with the following details:

- Score settings** | **Page settings**
- Paper**
 - A4 (210 x 297 mm)
 - Page width: 210.0
 - Page height: 297.0
 - mm | inch
- Orientation**
 - Portrait
 - Landscape
- Page Margins**
 - Left: 5.0 | Right: 5.0
 - Top: 5.0 | Bottom: 5.0
- Layout**
 - show Bar Numbers | Score size scaling (%): 100
 - show Page Numbers
 - Tempo Expression: Allegro
 - Composer: Composer
 - additional Text: Opus 1

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

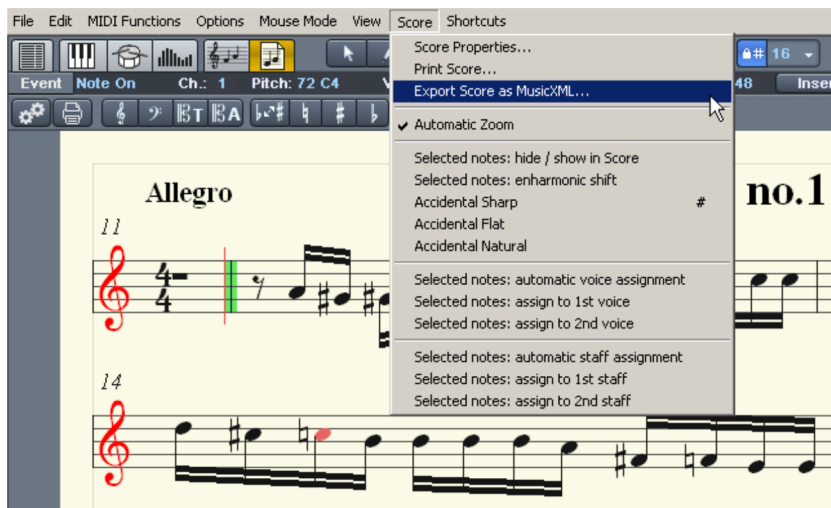
In complex arrangements, we recommend using two separate versions of a song for sequencing and for note print previews. This way, MIDI events for the note display may be changed in any way without influencing the playback. You may have to increase or decrease the length of a note in order to display 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.

MIDI editor shortcut keys

Apart from few exceptions such as "Space" for stop and playback, the keyboard shortcuts can be defined freely. Open the keyboard shortcut menu and mouse editor via the "Shortcuts -> Define shortcuts" in the MIDI menu. For some basic functions such as scrolling and zooming, all keyboard shortcuts from the VIP are used. You can define explicit shortcuts for all commands that are available in the MIDI editor menu.

Note: The mouse wheel settings from the arranger are also assumed, as is the switch for temporary zoom mode (special) which is preset as "Z".

Play/stop	Spacebar
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 + P
Delete selected MIDI data	Back, delete
Delete all MIDI data	Ctrl + Back, Ctrl + Del
Select next event	Right arrow
Select previous event	Left arrow

MIDI functions

Legato	Ctrl + L
Quantize notes	Ctrl + Q

Options

Scroll mode	F
Show event list	Alt + L
Show velocity/controller editor	Alt + V
Play clicked notes	Alt + P
Quantization grid active	Ctrl + G
Show quantization grid	Alt + G
Quantization options	Alt + Q
MIDI object editor	Ctrl + O
Audition panic-end played edit notes	Ctrl + F

Mouse mode

Selection	1
Draw	2
Drum (draw)	3
Pattern (draw)	4
Velocity change	5

Delete mode	6
Magnifying glass	7
Combine notes mode	8
Split notes mode	9
Mute notes mode	M
Select next event	Right arrow
Select previous event	Left arrow
Event pitch up	Up arrow
Event pitch down	Arrow down
Select next grid quantization value	Alt + Arrow down
Select previous grid quantization value	Alt + Arrow up
Select next length quantization value	Alt + Arrow right
Select previous length quantization value	Alt + Arrow left
Move play cursor forward	Page down
Move play cursor forward (fast)	Ctrl + Page down
Move play cursor back	Page up
Move play cursor back (fast)	Ctrl + Page up
Velocity controller combi-tool	Ctrl + 1
Draw velocity/controller freehand	Ctrl + 2
Draw velocity/controller lines	Ctrl + 3

Tempo Editing

Tempo and Time signature changes in the tempo map...

Tempo and Time signature changes are an important means of expression of your music. You can make these changes in Samplitude by using extensive options.

A precise beat grid is essential for precise editing. Samplitude offers you a convenient option for adjusting the beat grid to existing material (audio/video/MIDI).

Changes in tempo can be made using the marker or graphically using a **Tempo track**. **A tempo track contains the project's tempo map as an automation curve.** In principle each automation point of the curve corresponds to a tempo marker.

The tempo map displays all the tempo markers and determines the musical positioning for each time position. Put simply, the tempo map defines the beat grid.

You can adjust the **beat grid to certain time positions by using grid position markers** (e. g. existing audio or MIDI events). To do so, just drag a grid line of the beat grid to the desired time position.

It is even possible to tap the beat or entire bar positions. In this case, use the **"Grid Fit Mode" of the tempo map**.

Besides the **tempo map dialog** (view page 308), the **transport console** also serves as **optical control** of the tempo map and **tempo track**. The transport console displays the current values for tempo (BPM) and beat type at the current play cursor position.

In the **Tempo-Map dialog** (view page 308) you can specify which objects or markers in your position should be adjusted when tempo changes are made.



BPM marker, Tempo curve (>) and beat change (3/8) - under the tempo track.

Tempo markers

Tempo markers appear either as **BPM markers** or as **Grid position markers** according to the Tempo-map mode. **Time signature marker** (e. g. 3/4) are by contrast, independent from the Tempo Map mode.

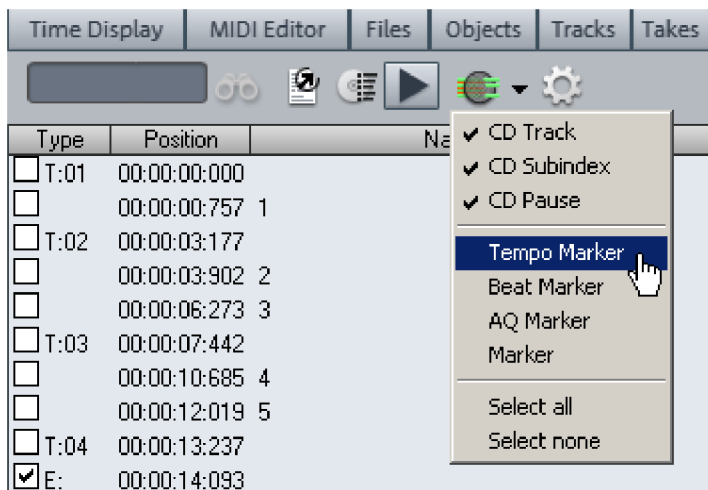
In total there are three types of tempo marker:

- Time signature marker (blue)
- BPM marker (red)
- Grid marker positions (green)

A time signature marker represent a time signature changes (e. g. from 4/4 to 6/8).

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 147).



Tempo Map Modes: BPM Mode / Grid Fit Mode

In the Tempo Map dialog (view page 308) you can choose between two modes: **BPM mode** and **Snap adjustment mode**. The selection can be accessed from the menu:

Play/Rec > Marker > Insert Tempo change

Play/Rec > Marker > Insert grid marker change

- In **Snap mode** you work exclusively with grid position markers and Time signature changes.
- In **BPM mode** you work exclusively with BPM markers and Time signature changes.



Modes can be changed at any point. The Tempo markers remain the same, but change their type. The effective Tempo map and beat grid remain unchanged. You can toggle between BPM markers and grid position markers at any time.

Example 1: Initially in the beat adjustment mode the beat grid is adjusted to a MIDI recording which contains lots of tempo changes. Subsequently certain musical tempo changes such as Rubato and Ritardando can be inserted for individual sections in BPM mode. (The tempo track can only be edited in BPM mode, in snap mode it serves solely as a display).

Example 2: The time signature of available grid position markers should be changed at the start of the piece, subsequently however the beat grid positions should be placed more precisely. As changing the time signature moves the beat borders and with it the grid position markers, change the markers to BPM markers because this will be unaffected by time signature changes. This is how you swap to BPM mode. Now change the time signature and go back to the beat adjustment mode in order to adopt the beat grid changes.

Tempo Change: BPM Marker

A new project first has a single "Master" tempo which can be defined in the transport console or in the project settings (shortcut: "I").

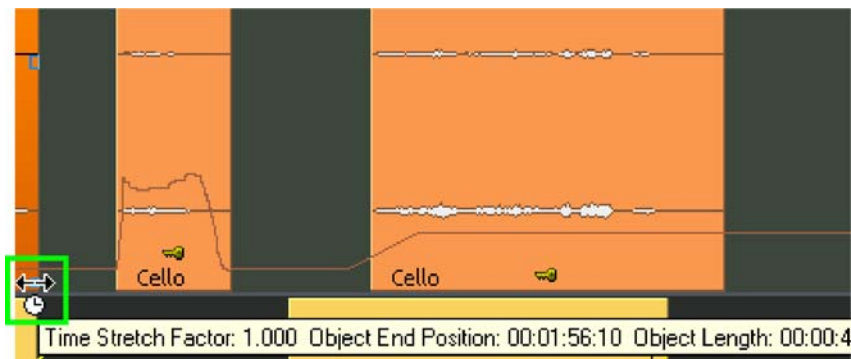
This Master tempo affects either the entire project or the start of the project up to the first BPM marker.

Exception: If the tempo should be interpolated to the first tempo marker, the master tempo is only effective directly at the project start, to then go linearly to the tempo of the next BPM marker.

You can make any tempo changes directly in the project window at any time or for any musical position (even between beats) using the BPM marker or corresponding tempo curve point.

The tempo markers are snapped to the next snap point when the grid is active. By dragging the tempo markers and holding down "Alt", you temporarily deactivate the grid function.

Note: Amplitude does not automatically adjust audio objects via tempo markers in case of tempo change. In order to change the length of audio objects to the defined tempo changes, use the pitchshift/timestretch mode and drag them to the end of the rear object edge of the audio object until this matches the original beat marking.



Time Signature Marker

Beat counter markers change the time signature from the marker position onwards, e. g. 4/4 beat to 3/4.

Therefore you can only insert time signature marker at bar borders and only position them at full beat borders.

Grid Position Marker

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 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.

Grid Tapping

You can set grid position markers in the tempo map grid adjustment mode during playback. To do so, assign the **menu command "Insert Grid Position Marker" a keyboard shortcut** of your choice, e. g. "P". Press "P" in time with the rhythm during playback and tap the beat grid or tempo map.

The following will happen: when you press the key, the nearest bar position of the selected beat grid will move to the current playback position and it will be marked with a grid position marker.

Tips:

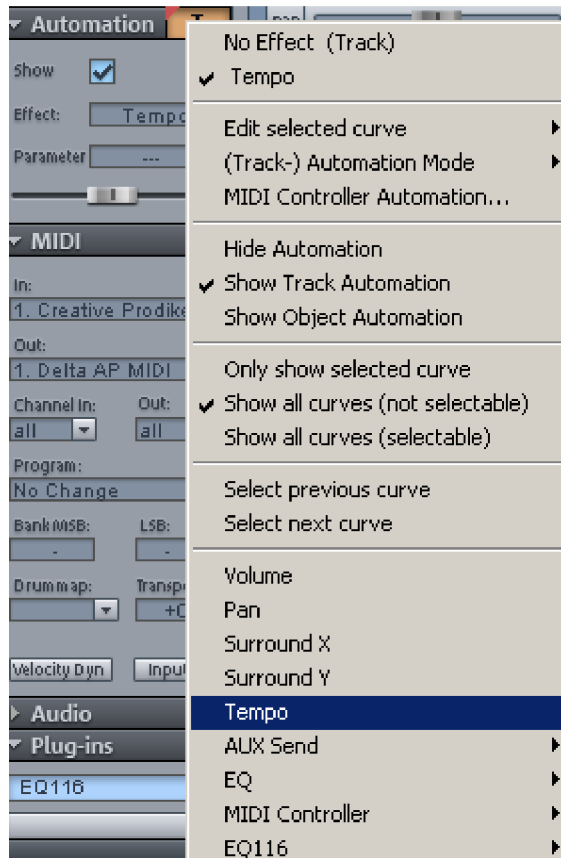
To detect the tempo, work in the bar grid, in entire bar borders first. In a second step in, you can detect slight tempo variations in the bar grid.

The BPM mode exceptionally allows you to set a temporary bar position marker while dragging with the mouse and the "Alt" key held down in timestretch mouse mode (or when loading older VIP projects).

The new command **Menu Edit > Tempo > Create click track** creates an audio track, which contains all metronome click as objects.

Tempo Track

The Tempo map is also available as an Automation curve "tempo" and can be graphically edited.



Note: Only one track can be used as a tempo track. The track that serves as a tempo track is freely selectable.

Changes to tempo markers or in the tempo track are automatically transferred.

The Tempo automation behaves differently to typical 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 (independent of the system option "Program > General > Snap also applies to curve automation points"). As usual the snap can be temporarily disabled using the "Alt" key.

Edit in the Tempo Track (only BPM Mode)

Insert tempo changes

- Double click on the tempo curve in **Universal mouse mode**:
- In the **Curve edit mode**: click (and drag) on tempo curve.
- Click in the **Automation draw mode**:
- **Ctrl key**: creates a Ritardando or Accelerando. A special BPM marker with tempo curves will be created before the tempo marker position ("Pre-BPM"), and, after the tempo marker a previous/new tempo will be used. The pre BPM value can be set with Ctrl + dragging.
- **Shift**: Creates a new tempo marker with tempo curve at the previous tempo marker.

Change existing curve points

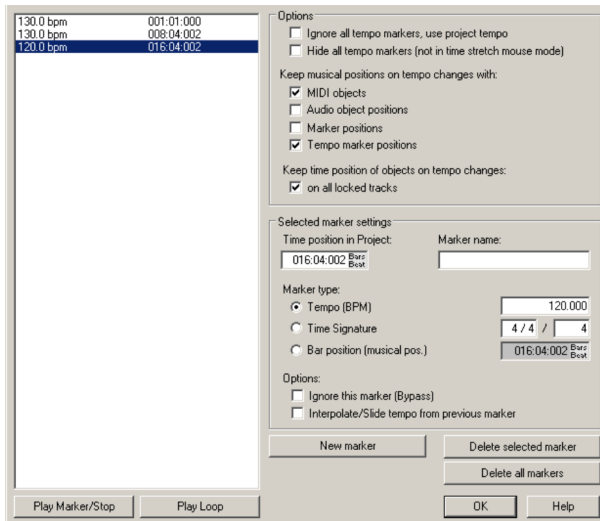
- BPM. value and positions can be changed by **clicking and dragging curve points**. For tempo levels: there are always two curve point for one tempo marker, one for position and tempo respectively.
- **Alt keys**: Temporarily disable snap. Toggle tempo changes and tempo curves as necessary.
- **Shift keys**: Only change and set BPM values (horizontal lock).
- **Ctrl + Shift + Click**: Switches between tempo curve and tempo step.
- **Right click**: open "Tempo and time signature" dialog (view page 310), to change values numerically.

Tempo Scaling / Change Global Tempo...

With this menu command all BPM values of the selected tempo curve points can be scaled with one adjustable factor.

Tempo Map Dialog

Set the playback cursor to the position at which the tempo or Time signature changes should occur. Now open the Tempo window from the Marker Menu (right-click on time bar) or from the edit tempo menu.



Click on the "New marker" button in the dialog. A marker is inserted at the current position. You can now use the marker option to set the properties of this marker. For a new tempo marker, enter the required tempo in BPM or the new beat for the time signature marker.

Please note 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 one has been created.

A time signature marker is always placed at the start of a new bar. If the play cursor is at a different position, the marker position is automatically moved to the beginning of the next bar.

Edit marker/Tempo map window

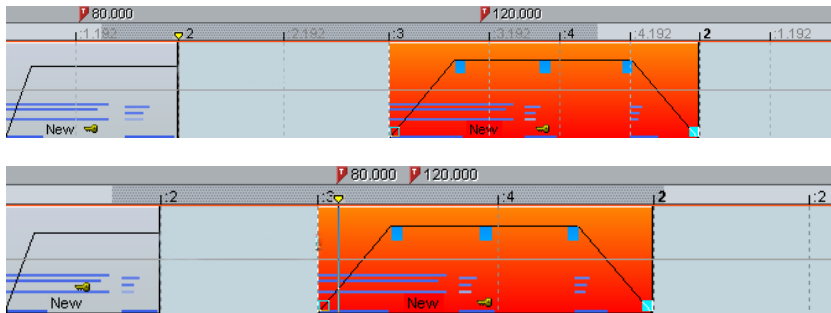
If a tempo marker is changed or a grid position marker is moved, this influences the following markers and audio/ MIDI projects in the VIP.

There are two main possibilities: The time position either remains constant, or the musical position remains and the time position is adjusted (Tempo/Time Signature window).

The time position in the VIP 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, its absolute time position in the project therefore changes. If the time position is retained the musical position changes accordingly.

When manipulating the tempo markers, the behavior of markers, audio, and MIDI objects is controlled separately. By default, the grid position of MIDI objects and tempo markers remains intact, and the time position of markers and objects remains the constant. The time position is always observed if you hold the "Alt" key when moving the mouse.



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 positions (03:01:00) were maintained.

If you want to keep the time position of objects, select the **"On all appointed tracks"** option

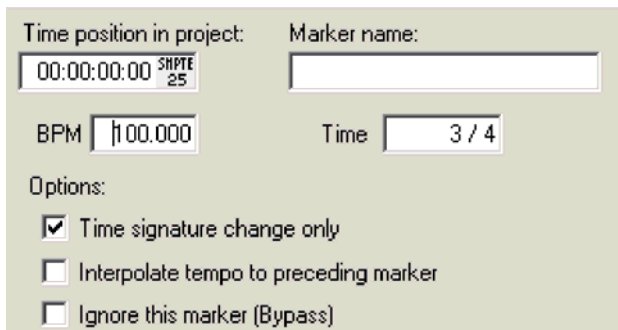
Additional options:

Each tempo and grid position marker can be ignored using the "Bypass" option. 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 these bar positions that originate from a previous tempo marker cannot be met by tempo interpolation.

Ignore All Tempo Markers, use Project Tempo

With this option you can ignore any previously created tempo markers in your project, so that only the project tempo observed.

Tempo and Time Signature Dialog



This dialog appears when you double click on a previously available tempo marker/curve point or if a new tempo marker should be set using a menu command. You can enter the desired tempo in BPM here (not for grid position markers) and, if needed insert a time signature change. Also you can choose here whether the tempo should be interpolated to the previous tempo marker (Ritardando or Accelerando).

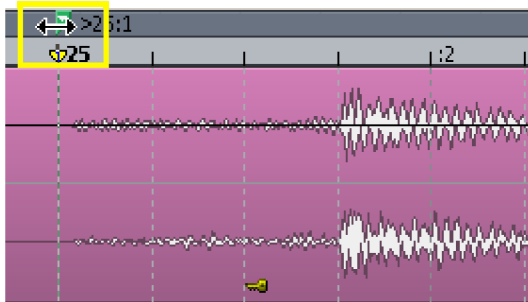
Manipulating the Tempo Marker in Timestretch Mouse Mode

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", and you can adapt this tempo by moving the mouse vertically while 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 create bar position markers with "Alt" + mouse click, and 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 is 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.

Software-Instrumente/VST-Plug-Ins/ReWire

Samplitude provides the option to integrate software instruments according to VSTi standards or via ReWire into a VIP, and these may be controlled via the internal MIDI functions and editors.

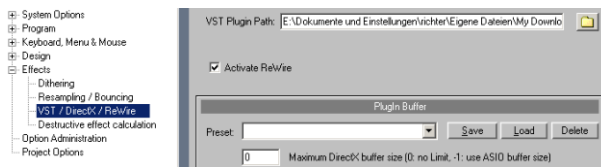
Software instruments may be accessed as audio track inputs and as mixer channels. In this way, the audio signals of a VST instrument may be edited further with all options featured by the mixer, including EQ, effects, and routing.

VST instruments offer sample-precise timing of MIDI events. The maximum number of plug-ins is only limited to processor performance. You can increase the number of applied VSTs almost without limit by using the integrated freeze function

Installing VST plug-ins

Install VST plug-ins according to the manufacturer's instructions. VST instruments and VST effects are not entered into the Windows registry, but must be saved in a certain directory.

Samplitude searches for installed VST instruments in the "VST plug-ins" program subfolder. VST instruments may be installed either in this folder or another path may be specified. If a VST plug-ins folder is already installed on the system, then please use the existing folder. The existing path must be specified in the system settings (shortcut: "Y" key) via **Effects > VST/DirectX/ReWire** (multiple paths may be specified). Simply enter one path after the other; each path will be scanned. Samplitude will automatically find new plug-ins that have been installed if they are present in the specified folders. If you add VST plug-in subfolders into the main VST folder (e.g. "Equalizer", "Filters", "Modulation", etc.), then these will be displayed as submenus when VST plug-ins are loaded.



The first time the program is launched, a search for existing instruments and plug-ins is executed (when the track settings are opened). In this case, not only are all the plug-ins imported, but they are also checked for their usability within the program. This search is only required once; the next time it is opened, the track settings will be opened immediately.

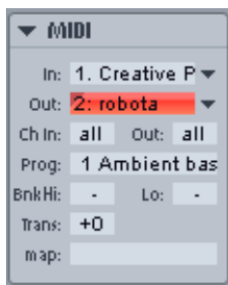
Tip: If you have problems with your current configuration, then you may reset all paths and installed instruments or plug-ins. To do this, delete the "VSTPlug-Ins.ini" file in the Samplitude folder. All paths will have to be newly entered. Plug-ins which were removed from the program prior to this will still be shown again.

Load software instruments

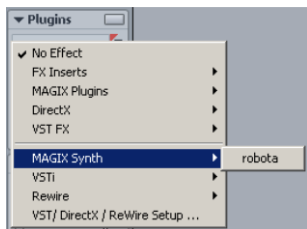
You can also assign a software instrument to each track; the instrument is selected as a MIDI playback device. Selected instruments or their individual outputs are shown directly in the first VSTi plug-in/insert slot of the track header and track editor and can also be muted (left click) and opened (right 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 audio section of the track editor:** "In" slot -> instrument outputs
- **In the playback device context menu,** (right click "Mute -> MIDI -> New instrument)
- **Plug-ins slot -> VSTi in the track editor**



Software instruments may also be created in the mixer:

- **Insert effect slot of the mixer -> VSTi**

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.

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.

Apply plug-ins at wave level

The menu item "Effects -> Plug-ins" lets you calculate the plug-in effects directly into the audio material. During virtual wave editing, the plug-ins will be loaded into the mixer's master area.

Detailed information about destructive editing is available in the chapter "Working in the project window -> Amplitude as a wave editor (view page 108)".

Load plug-ins at object level

Effects plug-ins may be applied at object level or via "Effects -> Plug-ins".

Effects menu: Note the check is not set for "Apply effects online". Select an audio object in the arrangement by clicking it. Next, open the "Effects" menu and select the entry "Plug-ins". In the dialog, you can load the desired plug-ins for the selected object one after the other via the "Add plug-in" button. Right clicking on the loaded plug-in opens its interface.

Object editor: Double click an object and select the "Object effects" view in the object editor. Under "Plug-ins", click the arrow to the right next to the plug-in slots. The plug-ins in the menu are available.

Load plug-ins at track level

At track level, you can load plug-ins in the mixer as well as in the track editor.

Track editor

- Select the track where you wish to load the effect.
- Open up "Plug-ins" if it is not yet visible.
- Now click on the little arrow next to the plug-in slot.
- Select any plug-in from the menu.

Mixer

- Open the mixer by pressing "M" or via the mixer button.
- Click the small arrow next to the insert slot for the desired track.
- Now select a plug-in from the corresponding submenu.

Load plug-ins at master level

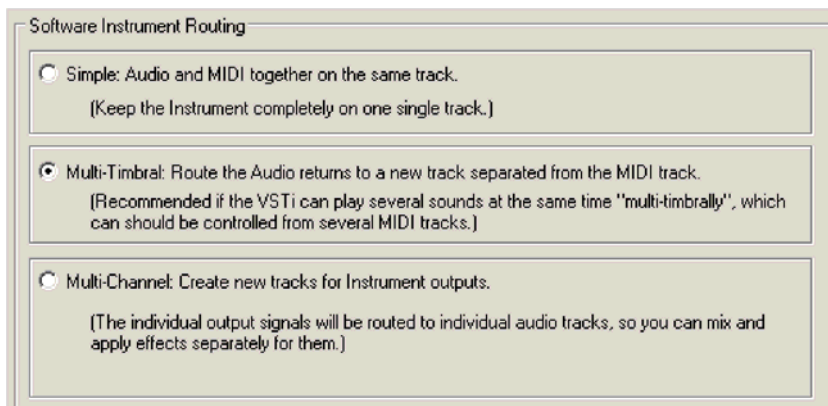
You can load plug-ins in the mixer's master section just like in the mixer tracks. Click the arrow to the right beside one of the plug-in slots and select the desired plug-in from the menu.

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). Amplitude 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

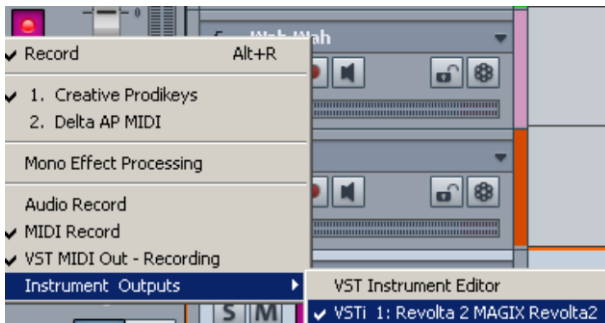
(for "Mixer FX monitoring/Hybrid Engine" only)

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 88) 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.

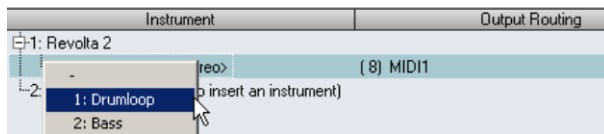
Routing VST instruments using the VSTi manager

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 may be easily routed by using the VSTi manager. Simply open the VST instrument to view the individual outputs, and then **right click the "Output routing" column**.



Now search for the track to be fed the corresponding individual output. The assigned output now appears in the target track's plug-in slot.

Alternatively, you may also **route directly to the track box of the arranger**:

To play the output of an already loaded (or instantiated) instrument on a specific track, **right click the record button** for the track, go to "Instrument outputs", and select the instrument output available in the project by placing a check mark. Several instrument outputs may be used simultaneously in one track. The outputs of several different instruments may even be combined into one track.

Deactivating the instrument outputs works the same way. Remove the instrument output by removing the check in front of the corresponding output in the "Instrument outputs" submenu.

Note that the routing options of the mix engine may also be used. A track containing the audio return of any instrument may be routed to any bus, AUX bus, or master.

Routing MIDI instrument inputs

Nearly all software instruments require MIDI data to be audible. To route the MIDI input to a specific track that contains the MIDI data, click the VSTi button in the track editor for the corresponding track, and then select the instrument in the "Out" slot that you want to activate. Previously loaded instruments are also featured in this list.

To open a completely new instance of an instrument which works independently of already existing instruments within the project, select its name from the "New instrument" submenu. Multiple instances of the same VST plug-in are indicated 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

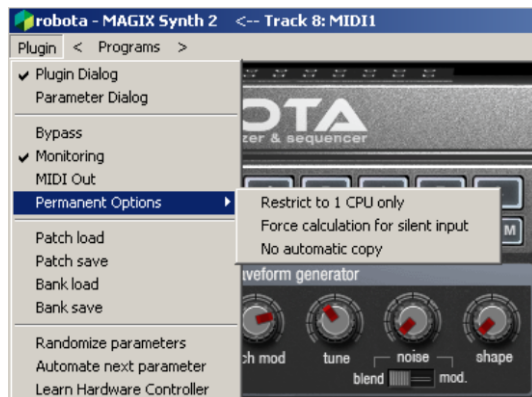
This approach is ideal for drum samplers to address an entire drum set and simultaneously mix and edit the drum sounds individually.

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**".



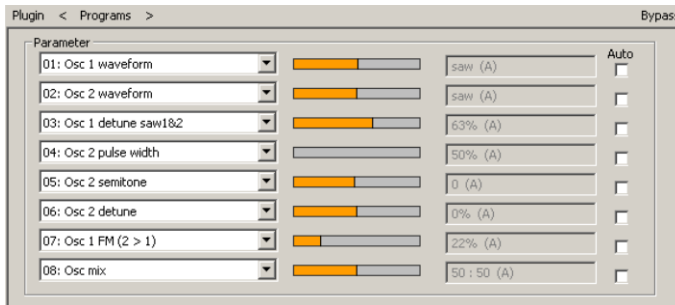
The graphical interface opens by default. The ReWire client application will open when ReWire instruments are selected.

"Programs" features the presets for the corresponding plug-in.

Besides "Programs", you will also find "Sidechain input (view page 324)" for those plug-ins that feature the "Sidechain" function.

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.

Plug-in menu functions

Bypass: Deactivates the instrument and mutes it. Note that some instruments use up processing power if "Bypass" is switched on. Remove the instrument if you no longer need it.

Monitoring: This option must be switched on to be able to hear the instrument when it is being played (live or recorded). If you want to have monitoring active when the track's REC button is on, select the "Tape monitoring" option in the "System/global audio" options dialog.

MIDI out may be activated for every track VST plug-in or VSTi in the "MIDI out" plug-in panel menu. This sends and records MIDI data from the VSTi 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 recommend only changing the "Permanent options" in case of compatibility problems. After making changes, reload the project.

Limit to only 1 CPU: All instances of VSTi will be calculated only on one CPU. This enables you to avoid a multi-CPU conflict between various instances of the same plug-in.

If you activate this option for an effect plug-in, all track or object effects on tracks using this plug-in will be calculated via a single CPU. As soon as you activate this option for an effects plug-in, the entire track and all effects will be calculated via only a single CPU.

Note: When using UAD cards, this option is activated by default.

Force calculation by silence input: When processing single tracks as economy tracks via the playback engine (view page 28), Samplitude normally doesn't process track signals if an audio signal is not present. This reduces the CPU load.

This plug-in option ensures that the complete track where the plug-in is embedded is calculated smoothly, even if there is no audio signal. This is especially useful for delay plug-ins in economy tracks or when bouncing economy tracks. This option is also recommended for all plug-ins that produce audio independently of their input.

Note: Continuous calculation takes place in the Hybrid Engine, even of empty or silent tracks. This function may be deselected in the global performance options by checking the option "Deactivation of empty or silent tracks for ASIO".

No automatic copying: This option prevents automatic copying of plug-ins when you split or copy objects.

Load/Save patches/banks: Save an instrument's settings and sounds. The standard formats for this are: *.fxp for patches or *.fxb for whole banks. Several instruments possess their own patch/bank format. Settings are then loaded and saved directly via the instrument's interface.

Set parameters randomly: Use this function to set all parameters of an instrument to a random value. For synthesizers, this option can cause surprising results, and it provides interesting results for sound design thanks to random generation of new sounds. Note, however, that parameters may only be set when they are accessible via the interface. With some very complex virtual synthesizers, e.g. modular systems, some parameters may not be modifiable using this option. Note that due to the random placement of various parameters, very extreme sounds may be generated. This may cause no sound to be produced at all or extremely loud volumes and frequency ranges. For this reason, keep an eye on the preview volume while experimenting with this function.

Automate next parameter: The next parameter which you change during playback will be recorded via automation. You may also do this by holding "**Ctrl + Alt**" while clicking and changing the parameters with the mouse.

Learn hardware controller: Activate this function to learn your hardware controller.

1. Grab and move the plug-in element that you want to learn
2. Move the element on the hardware controller you selected

Note:

The hardware controller element must have been learned in the hardware controller setup beforehand and may not be empty.

Learned elements permanently modify the internal mode in the hardware controller setup. Assignment is also available later for other projects.

Assignment may be cleared again via the "Restore modes" button beside the internal mode in the hardware controller setup

Detailed information about adding hardware controllers is provided under "Hardware controller -> Learn controller for plug-in/mixer elements (view page 372)"

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

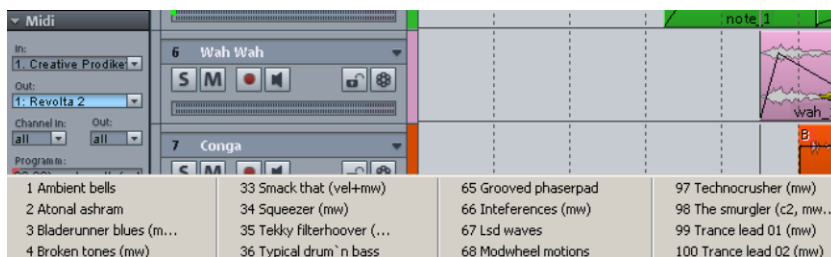
click on "REC" in the track editor and set a check next to the corresponding context menu item.

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. ammunition), 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 611)".

Freezing VST instruments

To freeze an instrument, mark the instrument in the **VSTi manager**, and select the **"Freeze"** option under **"Functions"**. After a short processing time, all MIDI and audio objects from the audio return track are replaced by a single 32-bit stereo audio object.

A frozen instrument will no longer be controlled by the track's MIDI files and can even be deactivated, provided that it is not used by other MIDI tracks.

Select "Unfreeze" in the VSTi manager to change the original MIDI data at a later point. The instrument returns to the state in which it was "frozen".

The "Freeze" function also works with single outputs. You can also freeze these tracks if they do not contain MIDI information and do not hold objects. The MIDI data that has been routed to the track or to the individual output of the instrument automatically creates an audio file that is played instead of the original instrument.

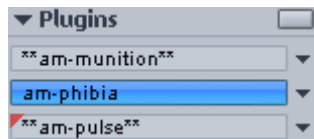
In order to avoid double playback of the data, manually deactivate the instrument's single output in the frozen track.

Note: To edit freeze data, you cannot use the "Track freeze edit" function only, since there is no MIDI data available on this track to be edited. Therefore, "unfreezes" the track and edit the send track's MIDI data. You can then refreeze the track.

"Inactive" state for VST instruments and VST effects

Plug-ins may be completely deactivated to free memory space. When reactivating the plug-ins the last state is restored in its entirety.

If you set VST plug-ins to "Inactive" by pressing **"Shift + Alt + Click"** in the plug-in slot of the corresponding track, these plug-ins are completely removed from RAM. This way, no more resources are used in conjunction with the hardware systems like "PowerCore" or "UAD".



The instrument on this track will be deactivated and removed from the disk. The "frozen" state of the instrument will be indicated using ** symbols on either side of its name.

ReWire Client Application

ReWire-compatible client applications (e.g. Propellerheads Reason) can be integrated into Samplitude as synthesizers.

Activate ReWire functionality in the system options settings ("Y") under "Effects -> VST/DirectX/ReWire". Afterwards, installed ReWire applications may be loaded as instruments into a MIDI out slot in Samplitude. All ReWire client applications appear as individual sections in the selection menu of the track's plug-in slot and are loaded as software instruments (VSTi). The client application should always first be started after launching Samplitude and closed again before Samplitude is ended. Some client applications may be opened automatically by right clicking on their name in the MIDI out slot.

The ReWire application may be controlled via MIDI, just like a software instrument. You can set up the ReWire client's individual output signals according to the multi-channel software instruments on multiple tracks. The client application runs, starts, and stops synchronously to Samplitude's time position.

The "classic" MIDI channel for MIDI notes and events is not important, it is replaced by the ReWire MIDI bus system. For each track, a MIDI object is able to control a ReWire MIDI BUS. This means that all of a MIDI object's events apply to the track for this ReWire bus, independent of the channel number the events originally had. Multi-timbral MIDI objects (like those created in the MIDI file import) cannot be played correctly with ReWire. However, you may access the ReWire client across multiple tracks on various ReWire MIDI buses.

ReWire supports up to 4,096 MIDI buses. A ReWire client only registers the MIDI buses that are actually available with the host (Samplitude). You can therefore select the bus within the track in Samplitude's MIDI channel menu (e.g. the receiving instrument for Reason).

Several ReWire clients may be opened directly via the host. If you start your client application manually, this automatically detects the presence of a host application and opens in a special client mode.

Detailed information is available in the section "Mixer -> Effects routing/Plug-ins dialog -> VST/DirectX/ReWire setup plug-in buffer (view page 176)".

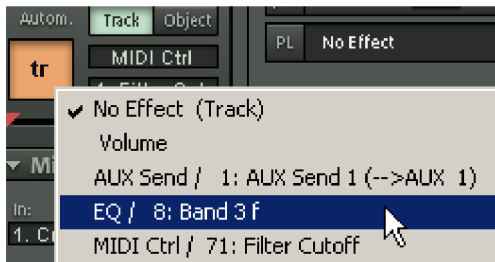
Automation

Samplitude provides lots of comprehensive automation possibilities at master, track and object level. You can enter automation movements on track and master level in realtime or draw them manually in the automation track. The created automation data are visible as controller movements in the channel or as a curve in the automation track. An unlimited number of automation curves are available on each track. The volume automation curve is set to +0 dB in the middle of the track. Upper track edge corresponds to +6 dB.

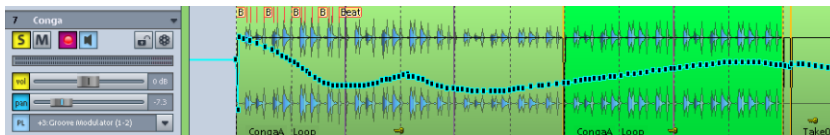
Select "Automation" from the menu or in the Track Editor by clicking on the appropriate button, to determine whether track or object automation should be displayed.



Equally you can select an automation parameter, in the "Automation" menu, or by right clicking on the Automation buttons in the Track Editor, which you can control with the automation slider in the Track Editor.



Alternatively, you can draw in the curve of the selected parameter using the pen in Automation Draw mode.



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.

Track automation - Basic procedures

Step 1: Select the track recording mode for creating automation curves for the individual tracks/channels. You can select either "**Touch**", "**Latch**", "**Overwrite**", and "**Trim**" mode for recording. "Overwrite" mode only applies to the last-activated automation curve, and "Trim" mode is intended for volume automation.

2. To automate a plug-in control element, select "**Automate next parameter**" in the plug-in menu, and click the desired function button in the plug-in dialog.



The arranger now displays an automation curve for this parameter.

You can also select the plug-in's parameter to be controlled by right clicking the automation display in the Track Editor.

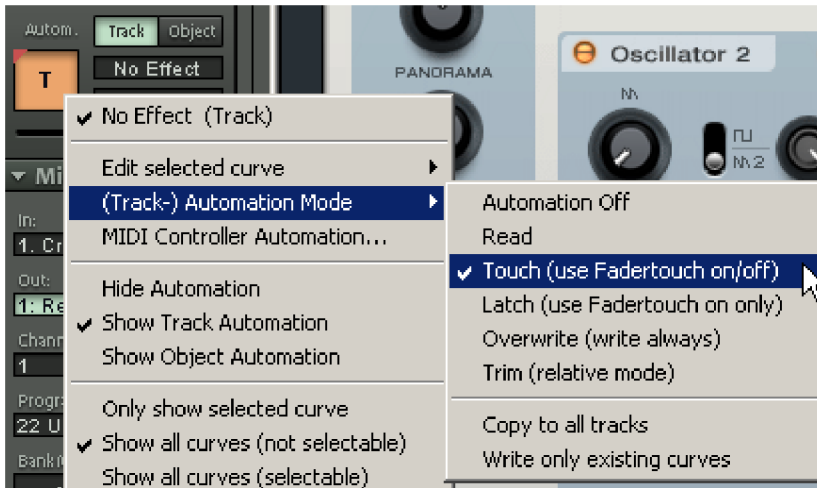
3. To start recording automation, press "Play". Whatever changes you made to the selected control element, will be recorded in real time during playback.



You can overwrite, edit and copy the automation data at any time. You can draw fine steps in the curve if you hold down "Shift".

Note: If you want to move curve points which were already created and selected vertically only, create a lock against horizontal movement by holding down "Shift". You can now move the curve points vertically in 0.01 dB stages. You can deactivate the fine grid by simultaneously holding down "Alt" in order to make rough changes.

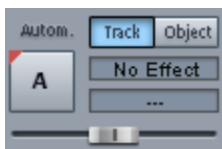
Automation modes



The automation modes define how automation data should be saved or displayed. Set the automation mode for each channel/track. By right clicking the automation button in the mixer channel, different automation modes may be selected. 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. Automation recording is stopped when you release the corresponding key. Once released, the control element will move back to the previously recorded position. You can set the fader return time in the **"MIDI controller automation"** dialog in **"Settings -> Time constant for user entry"**. The **"MIDI controller automation"** dialog can be accessed by right clicking the automation button in the track editor.

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 "Overwrite" mode, automation data is recorded right away, as soon as playback begins, independent of whether the fader is moved or not. Automation is written until playback ends or another mode is switched on. This way, you can quickly overwrite a previously recorded automation curve.

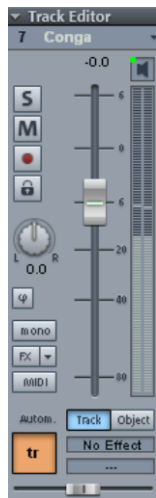
This mode is available only for the currently active, selected curve.

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, just like "Touch" mode (view page 330).

In Trim mode you can move pre-existing automation data. The channel's volume fader will firstly be set -6dB, 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.



When trimming a range, new automation points will be set at the range's 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 may 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.

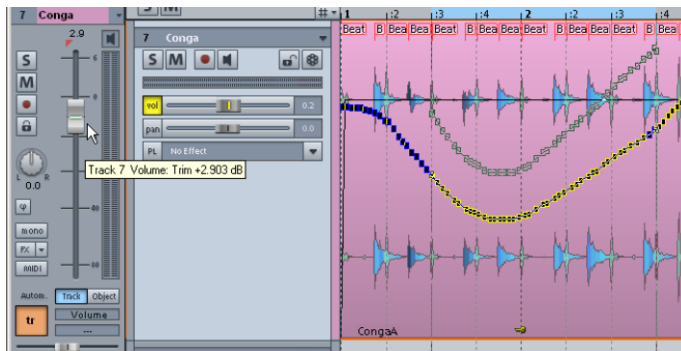
In "Trim" mode, automation data is only recorded as long as you touch the selected control element with the mouse or touch it from your external controller. As soon as you release the button, the automation recording stops. Once released, the control element will move back to the previously recorded position.

You can set the fader return time in the "**MIDI controller automation**" dialog in "**Settings - > Time constant for user entry**". The "**MIDI controller automation**" dialog can be accessed by right clicking the automation button in the track editor.

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

The option "**Transfer to all tracks**" applies the currently selected automation mode to all tracks/channels.

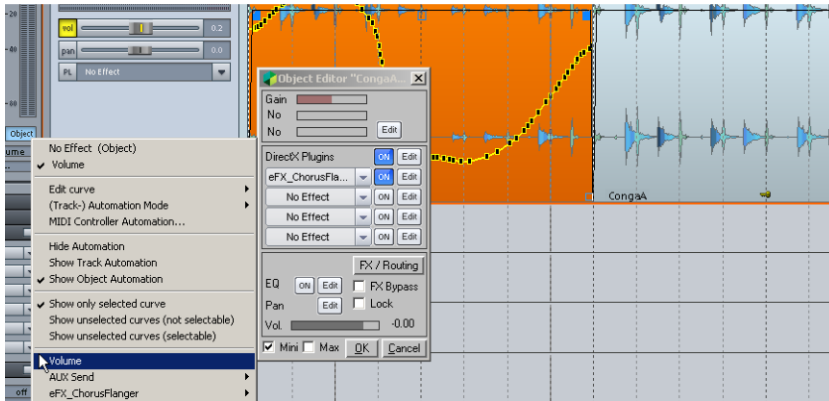
Automate existing curves only

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.

Object automation

At object level, you can automate the volume, AUX sends, and VST plug-in parameters.

To conduct the object-based automation, first select an object. Next, select "**Show object automation**" from the automation context menu and select an automation parameter in the lower section of the context menu. You can select object volume, object AUX sends, or any VST plug-ins embedded in the object for automation.



Switch to automation draw mode



and draw the object-based parameter changes directly into the object with the pen tool.

Master automation

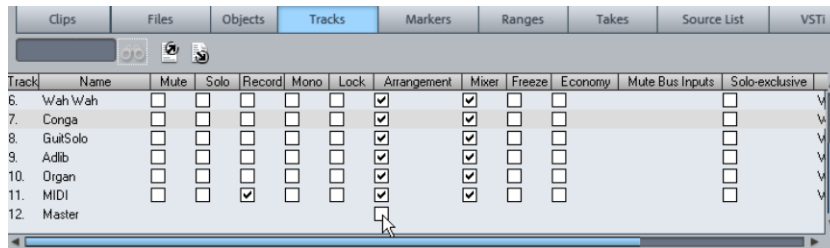
Volume, EQs, VST plug-ins, and MIDI controllers may be automated on the master level.

The master section of the mixer also provides automation buttons for volume automation of the stereo sum.

To make the master track visible in the arranger as well, right click the automation button in the master channel. Select the option "**Display master track in arranger**". The arranger will now feature a track outlined in blue and the name "Master". This track can also be automated in the arranger.

The master track may also be made visible via the track manager in the arranger window:

Open the track manager via the "**Manager**" button, and place a check next to "**Arrangement**" for the "Master" track name.



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

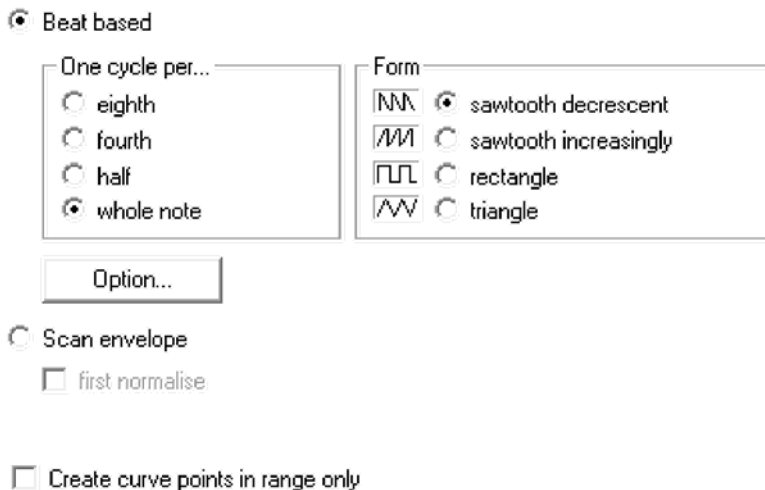
The following functions are now available:

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.

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.

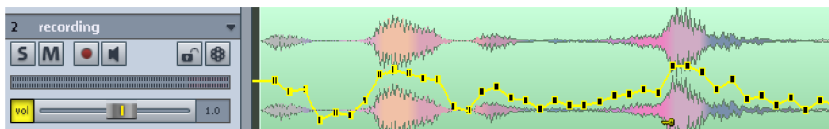


Beat-based: This option causes the form of the automation curve to follow the beat in the arrangement.

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.

Limit		Delay	
Minimum:	<input type="text" value="10"/> %	for minimum:	<input type="text" value="3"/> %
Maximum:	<input type="text" value="100"/> %	for maximum:	<input type="text" value="6"/> %
		shifting:	<input type="text" value="1"/> %

Shift moves the entire curve. This lets you create interesting off-beat effects.

Editing curves

Edit curve – Create

Opens the Curve generator (view page 335).

Edit curve – invert

This command inverts the activated automation curves.

Edit selected curves – thin out

While recording, the automation events are placed in very short intervals. The command "Thin out" reduces the number of curve points. The course of the automation is then displayed and reproduced even more accurately.

Edit curve – 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.

Edit curve – copy, paste, delete

The activate automation curve may also be copied, pasted into other tracks, or deleted.

If the option "Use snap for automation curve points" (System options -> Program -> General) is selected, automation points may be set while moving set values on the grid.

Delete all curves

This command erases all automation curves on each track automation or each object automation.

Track automation mode

Specify the track automation mode here. Detailed information about the automation modes is available in the chapter "Automation -> Automation modes (view page 329)".

MIDI controllers

This launches the MIDI controller automation (view page 338) dialog.

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.

Display object automation

If you select this display option, only the object automation will be displayed for all tracks.

Display selected curves only

Views only the selected curves for the corresponding track. This helps keep an overview if several automation curves have been created.

Show unselected curves (not selectable)

The unselected automation curves are also shaded, but may now be activated with the mouse.

Show unselected tracks (selectable)

The unselected curves are also shaded, but may now be activated with the mouse in universal mode/curve editing mode/automations drawing mode.

The last section of the context menu lists the ranges that automation parameters are available for when automating the corresponding track or the corresponding object.

Curve generator

This menu item opens the curve generator.

MIDI controller automation

Automation/controller

For each track, you may also draw MIDI controller curves automatically and send them to a VST plug-in or MIDI synthesizer over the track's MIDI output device. You can assign a MIDI channel to each MIDI controller curve.

Open the MIDI controller dialog for the corresponding track by opening the context menu (right click on the automation display in track editor), and then selecting the "MIDI controller automation".

Shortcut: Ctrl + Alt + A

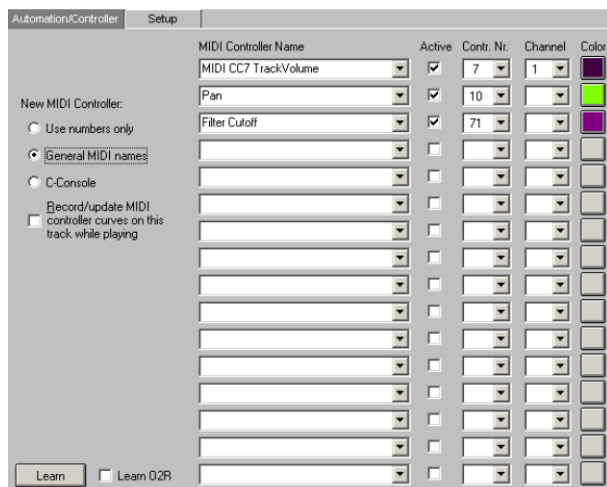
"MIDI controller name" or "Contr. no." selection menus for all available MIDI controllers for the selected track. These MIDI controller slots may be set up individually for each track.

A MIDI controller may be assigned to a free slot in the name field according to the general MIDI name specifications (e.g. volume, panorama, or LSB bank change) or according to its number (0-127) in the number list box.

If you select "MIDI controller (according to general MIDI nomenclature)", you will see the list of controllers according to the GM standard in the "MIDI controller name" field selection menu.

For the external C-console program, a pre-defined name list is also available for selection.

- MIDI controllers (sorted by numbers)
- MIDI controller (according to general MIDI designation)
- MIDI controllers for C-console



- Active: Switches the controller curve on/off
- Controller no.: MIDI controller selection
- Channel: MIDI channel to which the MIDI controller sends values
- Color: Color selection for each automation curve

By selecting the option "Record/update MIDI controller curves to this track during playback", an automatic recording of selected MIDI controllers curves for this track takes place.

To fill an already occupied automations slot, simply enter another controller name or another controller number.

Delete an automations curve by selecting the topmost (empty) entry in the "MIDI controller name" field.

Settings

Detailed information on automation curve settings and MIDI controller can be found under "Automations/MIDI controller settings (view page 343)".

VST plug-in/VST parameters dialog

All of the parameters that may be automated may also be selected and activated in the VST plug-in's parameters dialog.

This allows automation to be recorded directly via the selected parameter controllers during playback.

You can open the parameter dialog in the plug-in's menu via "**Plug-in -> Parameter**".

Detailed information about the VSTi parameter dialog is provided in the chapter "Software/VST instruments -> Plug-in parameter dialog (view page 320)".

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.

"Draw" mode

Select "**Draw automation curves**" mode to draw automation curves for a parameter or a MIDI controller.



Drawing track automation

First, select a control element in the track editor/mixer or VST plug-in using the key combination "**Ctrl + Alt**".

Draw the automation curve for the selected parameters straight into the selected track using the pencil tool.

To create an automation curve using the pencil tool, draw the corresponding automation curve by dragging with the mouse cursor on any position within the track.

Drawing object automation

- Switch to "**Object automation**" mode in the track editor
- Right clicking the automation area in the track editor opens a context menu; the lower section displays all available object automations. Select the desired automation parameter (e.g. AUX Send 3). A dotted, inactive automation curve appears in the object (hint: newly created, inactive "AUX send" curves are located on the lower edge of the object).
- Next, draw the automation for the selected parameter in the selected object by holding down the mouse button.

VST parameters are shown in % while drawing; volume and "AUX send" parameters are displayed in dB.

More information about "Automation draw" mode is provided in "Screen elements -> Toolbars -> Mouse mode bar -> Automation draw mode (view page 71)"

Edit automation curves

Besides the normal mouse (create or delete curve point), a special "Curve edit" mode or "Object/curve" mode



or "Object/curve" mode



is provided.

In **Curve edit mode**, you may directly draw, move, and delete curves with a double click. The mouse cursor switches between a **hand or moving tool**. If the mouse is not directly above the curve, you may select an area of automation points using the **calculation tool**.

In **Object/curve mode**, you may also create new points on the curve, move, and delete curves with a double click. If you move away from the curve using your mouse, you will automatically switch to "Object" mode to move objects and edit the start & end position, the fade-in & fade-out phases, and the object volume.

Detailed information about curve modes can be found in "Screen elements -> Toolbars -> Mouse mode bar"

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/MIDI controller settings

In the "Automation curve" dialog ("**Ctrl + Alt + A**"), you will find an additional subdialog "**Settings**" for configuring all automation settings. There is a range of "Global settings" that affects all tracks equally.

MIDI resolution: Time constant (in milliseconds) for sending current controller data. When starting playback, the controller data is transmitted after this time has elapsed if its value changes (within the frame of 7 bit MIDI resolution). During playback, the MIDI controller data is retrieved and sent to the respective MIDI device.

Time for entries by users: As soon as a controller curve is updated (i.e. new controller data for this curve is received) previous data (of this curve) is no longer transmitted. If no further data is transmitted, Samplitude transfers the original data after this time expires.

Note: This sets the fader return time for the write automation modes.

Visual refresh time: The frequency of redrawing can be adjusted using the "VIP refresh time" input box.

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.

Send MIDI curve data: Activates or deactivates global transmission of MIDI controller data.

Show received data automatically: The curves in the project window can be updated automatically. The refresh rate can be adjusted via the "VIP refresh time" window.

Automatic track allocation of the new MIDI controller

Set to what extent new MIDI controllers should be automatically allocated to tracks here.

The following options are available:

- None
- Only tracks with "Recording/update MIDI controller" activated
- All tracks with the same input device
- Current track + MIDI channel (C-console mono tracks)
- Current track + MIDI channel (C-console stereo tracks)

Detailed information about the first tab of the dialog ("Automation/controller") is provided in the section "MIDI controller automation (view page 338)" above.

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 timecode, or SMPTE timecode 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 timecode 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 timecode. 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 timecode 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 351).

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.

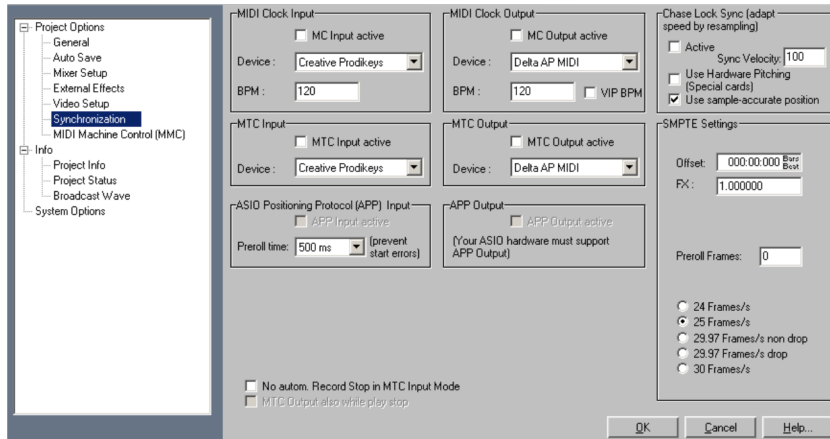
Note: Inform yourself regarding about whether your audio device and the ASIO driver support the APP protocol.

Synchronization setup dialog

The settings options for synchronization are available via the synchronization setup dialog.

Menu: File > Program preferences > Synchronization setup

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 775)".

No automatic record stop in MTC input mode

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 timecode is received again, Samplitude will begin playback. In any case, if you want Samplitude to record again when receiving MTC, then activate this option.

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 33).

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).

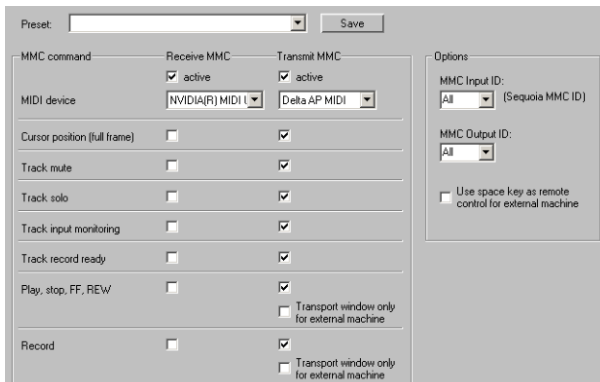
MIDI Machine Control (MMC)

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 control no longer controls playback and record directly within the program, but rather the external MMC device. Play, stop, fast forward, fast rewind, and rewind no longer have a direct effect on playback in Samplitude. 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.

More 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: Enter the input ID of the MMC device to control remotely.

List of implemented MMC commands

Supported receive commands

Full frame	
Locate	44
Play	02
Deferred play	03
Stop	01
Record pause	08
Pause	09
Record	06
Rewind	05
FFWD	04
Shuttle	47
VARIABLE PLAY	45
READ location	42 01
READ record status	42 4D
READ track record ready	42 4F
READ track record status	42 4E
WRITE track record ready	40 4F
READ motion control tally	42 48
READ track mute	42 62
READ track input monitoring	42 53
WRITE track mute	40 62
WRITE track input monitoring	40 53

MAGIX MMC extension

F0 7F <device_id> 06 6F <...> F7

Set marker	6F 01	New auto numbered marker
Navigate play cursor 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

Send commands

Full frame		
Locate	44	
Stop	01	Transport control remote mode
Play	02	Transport control remote mode
Rewind	05	Transport control remote mode
VARIABLE PLAY	45	Transport control remote mode

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 many 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.

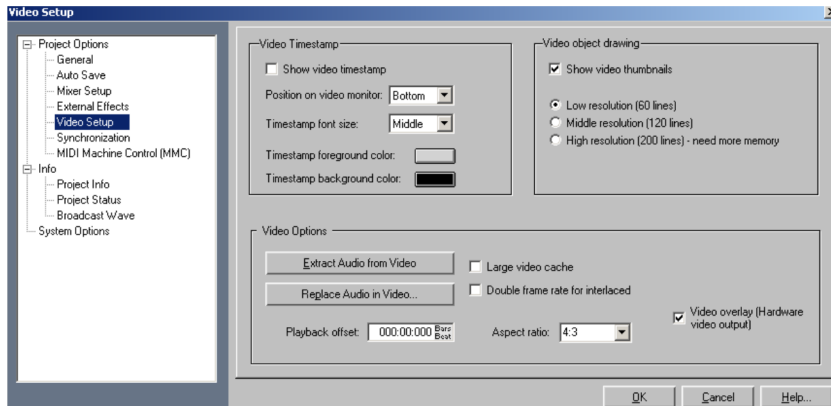
Important: Always make sure that the sync start position of the slave system is identical with the start position of Samplitude. If SMPTE sync start cannot be set for the other sequencer, you can also enter the starting time of the other system in Samplitude in "Synchronization -> SMPTE settings -> Offset".

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**.

Media link/video setup

The integrated video engine allows video objects to be added to the first track. Various video files like AVI, DV, MOV, MPEG, MXV may also be loaded. WMV or image files such as BMP or JPEG may be loaded into the arranger window via drag & drop or via "File -> Load/import -> Load video file".



Please ensure that the video files included in a project are identical in terms of dimension, aspect ratio, and frame rate. If possible, use only videos that work only with keyframes and without interframes (DIVX or MPEG are not very suitable).

The video soundtrack is loaded into the project, MPEG-2 sound is then converted into WAV format.

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, it is good to keep in mind that these will be loaded completely into RAM and not successively read from the hard disk.

Right clicking the video object enables you to

- open the video setup dialog window
- show/hide the video window
- replace the video file
- move/shift the video object by entering values

Video options

Show video time stamp: This option opens the time stamp display in the video monitor. You can choose for it to be displayed at the top, center, or at the bottom.

Video object drawing: The option "**Show video thumbnails**" displays thumbnails in the video track in different resolutions. Improve system performance by deselecting the box to shut off the video object display completely. The video file will continue to be played in the

video monitor. VIP draw mode 2 ("Tab" key) may 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.

Aspect ratio: Select an aspect ratio of either 4:3 or 16:9.

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.

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).

Hardware Controller

Hardware controller - Introduction

Samplitude lets you easily control the software using external hardware controllers. Templates (.cps files) for advanced controllers come supplied. The number of supported devices is increased continually and they are adapted to the new devices that appear on the market.

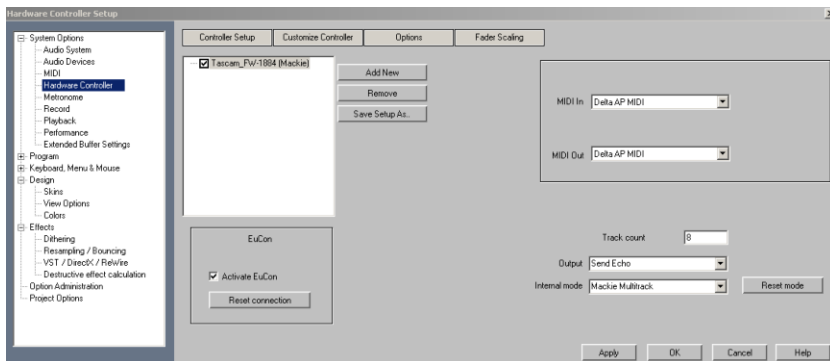
A MIDI interface is required to use a hardware controller. Please ensure that the program's MIDI functions are not deactivated, (Menu, "File > Program Preferences > System options > MIDI > Deactivate all MIDI functions") since they are necessary for addressing the MIDI controllers.

The hardware controller setup contains four registers.

- **Controller settings:** Selects your controllers and assigns MIDI ports.
- **Adjust controller:** Learns or changes functions.
- **Options:** Carry out additional individual settings.
- **Scale faders:** Adjusts faders to the scales on the controller.

Hardware controller settings

Access the "**Hardware Controller Setup**" dialog by pressing "Y" on your keyboard. The "System Options" dialog will open; click the "Hardware Controller" subcategory.



1. Click the "**Add New**" button in the controller settings. Select your controller or a compatible device from the list. Pressing "**Remove**" or "Del" on your keyboard will remove the selected controller. Save your individual configuration by clicking "**Save as**".
2. Drag & drop to modify the order of controllers and their internal assignment. A controller that appears beside another one corresponds to an expansion module. Click the check box above the controller to activate/deactivate a controller temporarily.
3. Select the **MIDI In and MIDI Out ports** that your controller is connected to on the right.
4. Close the dialog. The controller is now operational.

Using multiple controllers, extenders, and multi-bank controllers

A **maximum of four controllers** may be positioned in the controller selection list. For example, if you select a Mackie Control and want to add an extension to it, then just add a Mackie control again. 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 controller, too.

If the order of both devices is not correct, simply swap their positions via drag & drop or change the MIDI ports.

If you wish to have both controllers working parallel and independently, drag them using the mouse so that one is directly beneath the other in the list.

Activate EuCon: Activate this check box if you are working with a EuCon controller. The connection may be updated at any time via the corresponding button.

Track count: 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, there are also controllers that have a native mode that is able to learn several channels (64 max.). This value is 1 for single track controllers.

Output

Deactivate: If your controller does not have any motorized faders, you can use this option to deactivate transmission of controller data.

Normal Send: Program data (e.g. mixer fader changes) are sent to the controller. MIDI controller data is received and processed.

Send Echo: Program data (e.g. mixer fader changes) are sent to the controller. MIDI controller data is received and processed and sent back to the controller. This is required by the controller to update features such as LED displays and buttons. If your controller has difficulties with this (e.g. Logic Control), activate "Normal Send".

Send Echo (without fader touch): Program data (e.g. mixer fader changes) are sent to the controller. MIDI controller data is received and processed and sent back to the controller. In this case however, the fader values are not returned when the faders are moved (activated fader touch), but rather only the last value after the fader is released is sent (in order to prevent increased data volumes). If your controller experiences difficulties with this feature (e.g. Logic Control), then activate "Normal Send".

Hardware controller – "Internal" mode

The internal modes are used to adapt to different hardware controllers which operate using different concepts. Depending on the mode, the controller display is controlled differently and Samplitude interprets the controller information differently. The effect is that a particular physical control element controls different parameters in Samplitude, depending on the internal operating mode.

The function "**Learn hardware controller**" in the mixer menu or in the menu for any **VST plug-ins** (view page 372) enables these data structures to be manipulated. In this case, the active operating mode may 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.

To restore the original configuration, press the button "**Restore modes**".

Note: If you are using a hardware controller that is not listed in the hardware control setup, then select one from the list that is most similar in concept and design.

Following internal modes (Mapping modes) are available:

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 AlphaTrack

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 v pots. The display page is toggled with the keys "Mixer track up/down".

Frontier Tranzport

Functions: For transport only

JLCooper (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.

Preonus fader port

Active control

The currently active mixer element (recognizable by red highlighting) is controlled by the first fader.

Mackie HUI

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.

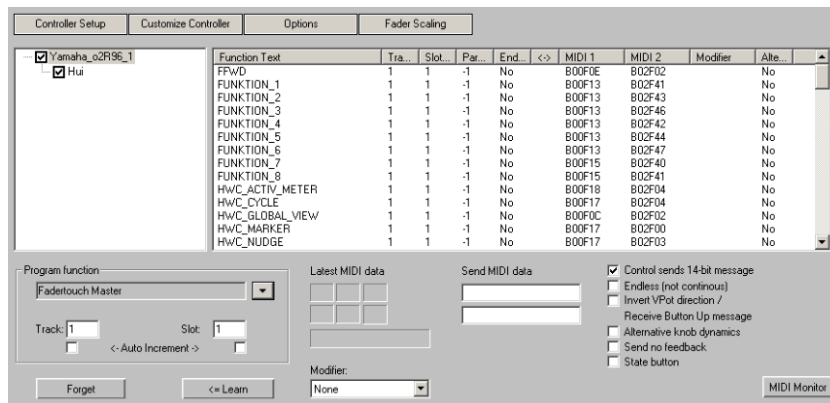
Customizing controllers

Most devices available on the market understand protocols such as Mackie Control or HUI. If your device is not included in the list (view page 373), simply select the nearest compatible device.

Note: Before adapting an existing template to your controller, save it under a new name via "Save as..."

If you can't find your device in the list or want to change the default settings, you should proceed as follows:

1. Open the "Customize controller" tab. The following dialog appears:



2. Use Mackie Control if the controller is able to emulate this mode. In this case, 7-bit messages are transmitted. This is also visible in the MIDI input monitor. Pressing a button displays a new line; another line is displayed upon release. HUI receives 14-bit messages, i.e. two lines respectively.

3. In the mixer, which should be open at the same time, you can now select the element you wish to program on your controller. The saved function is displayed in the section "Program function", e. g. "Fader master" for the fader in the Samplitude mixer. Alternatively, you may also open the context menu via "Program function". This contains a list of all remotely controllable functions in Samplitude. The uppermost commands are purely mixer functions and are suited solely for the controller connection. The lowermost commands are illustrations of the complete Samplitude menu structure. You can use this to assign each command from a menu to a controller key. Please note that when working with menu commands, the key LEDs of the controllers remain inactive.

Here are a few special learnable program functions:

- Mixer: Global track arming/mute/solo
- Mixer: Global FX mute button
- Mixer: Monitoring volume
- Mixer: Monitoring AFL/PFL button
- Mixer: Flip mode (exchange faders and vpots)
- Mixer -> Navigation: Learn AUX modes button for AUX1... AUXx
- Modifier: SHIFT lock (e. g. for JL-Cooper MCS-3800)

1. Select a program function via the context menu or by moving a mixer element.
2. Now move the fader or press the corresponding key on your controller. The MIDI data for this element is displayed in the section "**Latest MIDI data**".
3. Press "**Learn**" and the new data will be assigned to the function in Samplitude.

Note: Hardware controllers with touch-sensitive faders send a message whenever they are moved and released (FaderTouch). "Learning" must be activated before moving and releasing a fader.

If you want to learn a key/fader for multiple tracks, e. g. mute, select the corresponding function in "Program functions" (e. g. "Mixer -> Tracks -> Track mute"). Now activate "**Track -> Auto increase**" and start programming the first track. After pressing "**Learn**", the value of the track is automatically increased by "1". Simply move/activate the next element on the controller and press "**Learn**" again.

Use the "**Delete**" button to remove assignment of a function.

If a controller moves in the wrong direction after programming it, repeat the programming sequence and activate the "**Invert vpots direction**" option.

Last MIDI files: These fields display the last used MIDI files.

"Send MIDI message This function can be used to: send another MIDI message to the controller for LED control.

MIDI messages that are sent by a hardware controller's control element are normally the same messages used by the application to control the control element remotely. This may be the position of a fader or a key's LED light, for example. There are also controllers that expect different MIDI messages. These may be entered separately in the fields.

Control sends 14-bit message: Activate this button if the control element to be learned sends two MIDI messages. HUI protocol does this, for example. There may be different control elements on a single controller, in which case several send 7-bit messages while others send 14-bit messages.

Endless: Activate this option if the control element to be learned sends relative movement (e. g. dials to the right or left) - in contrast to absolute positions.

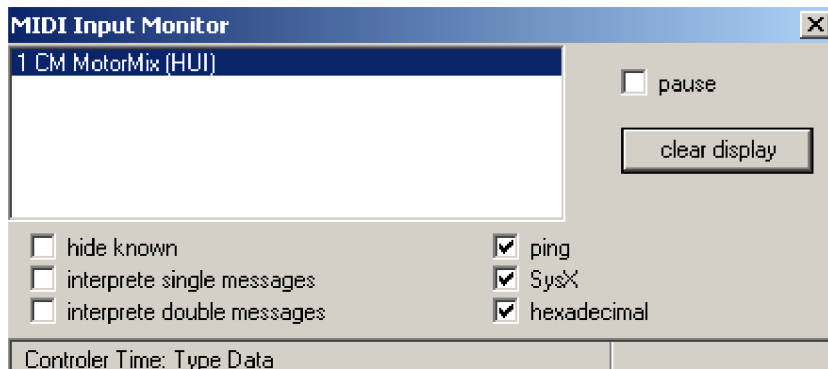
If a controller moves in the wrong direction after programming it, repeat the programming sequence and activate the "**Invert vspots direction**" option.

Alternative controller dynamic: If a dial makes overly large 'jumps' when moved quickly you can edit that here.

Don't send feedback: Regardless of the output mode, you can specify the controller settings for each controller, e.g. if feedback should be prevented from returning to the controller.

Status key: Controllers normally send a MIDI "IN" message when a key is pressed and then a MIDI "OUT" message when the key is released. Some controllers send alternating "IN" or "OUT" messages by pressing and releasing respectively. To be able to use keys in this fashion, activate this message during MIDI learning.

MIDI monitor: The MIDI monitor can be opened using the button of the same name. It displays all MIDI files, which are sent by the Hardware Controller Module to Samplitude. As sources of sent MIDI files, Samplitude's MIDI Editor as well as activated plug-ins come into consideration.



If you press the "Shift" key when clicking a button at the same time, the MIDI monitor will remain open once you close the hardware controller settings.

If you click a button while holding down the "Ctrl" key, then you will be able to move the MIDI monitor as an independent window freely on your monitor.

Of course, both modifiers ("Shift" + "Ctrl") may also be held down together to connect there functions.

Modifier: Assigns modifier keys for the computer keys (Shift, Alt/Option, Ctrl/Control) to execute special functions. Pressing the assigned modifiers will execute these special functions.

The "Shift", "Option", and "Control" modifiers are available for hardware controllers. To be able to use these modifiers, they will need to be learned first as program functions on the hardware controller keys. To learn a function with modifiers to a key, you need to set the

desired combination of modifiers in the modifier field. Of course, not all functions may be used with modifiers.

Other learned functions: Learned functions are displayed clearly in a list. Click on an entry in the display window and the setting will be transferred to the dialog element. Individual changes may now be written over older entries by pressing "**Learn**".

Controller elements

The hardware controller elements featured in Samplitude provide additional options:

- Update locator at initialization
- Track monitoring is no longer coupled to record ready, but is rather individually learnable
- Optimize peak meter processing
- ActiveControl mode available for plug-in controls/VSTi
- On-the-fly mixer mapping also for buttons
- Hardware controller follows the mixer while scrolling in the mixer with "Ctrl" held down
- Controls and automation for plug-in parameters via vpots or faders
- Automation with endless controllers (with and without modifiers for Pan, AUX, and EQ)
- Automation button in "HUI" mode glow red when automation is being written or green when automation is being read (controller-dependent)
- VPot controls with depressed "Shift" for gradual controller changes

Hardware controller options

Depending on the controller in question, received MIDI messages will be interpreted differently. In the "Options" tab, adjust and individualize settings for different controllers and operating methods.

The screenshot shows the 'Options' tab of a software interface. On the left, a list of controllers includes 'Mackie HUI' which is selected. The main area contains several settings:

- Protocol: HUI
- Display Mode: HUI
- Locator Mode: HUI
- Peakmeter Mode: Tascam
- Bank switch track number: 8
- MIDI ping to controller: 900000
- MIDI ping from controller: 90007?
- MIDI ping interval (ms): 1000
- Display interval (ms): 100
- Locator interval (ms): 100
- AnySolo LED: 000000
- Manual Hardware ID: ff
- Request Hardware ID:

On the right side, there are several checkboxes for advanced options:

- Ignore SysX input data:
- Disable SysX out:
- Fader touch selects track:
- No fader update after release:
- Controller follows selected track:
- Selected track follows controller:
- Mixer follows controller:
- Mixer follows controller on track selection:
- Show track numbers as track names:
- Activate assignment display:
- Control VPot LEDs:
- Process messages on open dialog:

Protocol: The protocol specifies a series of internal interpreting parameters. You may select Mackie Control, HUI, JLCooper, and Tascam.

Display mode: Many controllers feature LCD displays which differ in size and number of lines, and which also need to be controlled differently. For controllers without a display, you can deactivate the display control. You may select Mackie Control, HUI, and JLCooper.

Locator mode: Several controllers possess a special display for the current time position (locator). Locator mode may be set in the following display types: Off, Mackie Control, HUI, and JLCooper.

Peak meter mode: Some controllers possess a special peak meter display. "Peak meter" mode sets the way that the display is controlled. You may select Off, Mackie Control, HUI, JLCooper, and Tascam.

Bank switch track number: A project in Samplitude can include more tracks than are available to the controller. In this case, the controller only displays a section of the available tracks. To change this section, change the channels or the bank. While switching channels changes the section exactly to the track, a bank switch can jump several tracks. Depending on the number of tracks on the controller and your preferred working style, you may set different step intervals per bank.

MIDI ping to controller: Most controllers expect a periodic incoming MIDI message (ping) to ensure that there is communication with the application. Different messages may also change a controller's operating modes. Enter the message to be sent to the controller.

MIDI ping from controller: As a response to ping messages, controllers usually send a ping message. So that no confusion results, you may enter these explicitly so that they may be ignored by the application.

Display interval (ms): Display control occurs in certain time intervals and not continuously. This limits the amount of display information. A small interval accelerates the display on the controller, but enlarges the amount of MIDI data that needs to be transferred.

Locator interval (ms): Locator display control also takes place in intervals. If the time position within an interval changes, only the current value is sent to the controller after it expires. Reduce this interval and the locator display will react more directly.

AnySolo LED: Some controllers possess an LED which indicates that at least one track in the project has an active "solo" function. Since no button belongs to this LED and it is not learnable, you may set the corresponding MIDI message here.

Manual hardware ID: To activate "Mackie Control" display mode, an additional parameter is necessary: the hardware ID. This may be automatically queried with new firmware versions or entered manually.

Query hardware ID: Activate or deactivate the automatic hardware ID query. Deactivate this function if you aren't using the "Mackie Control" display mode. This helps to avoid incorrect interpretation of the query message.

Ignore SysX input data: SysX data sent from the controller won't be processed by the application.

Don't send SysEx: Samplitude sends different messages via SysEx. What is sent exactly depends on the selected display mode, locator mode, etc. If problems occur with SysEx messages while using a controller, then you can use this to stop sending these messages to the hardware controller.

Fader touch activates track: This option activates the track that belongs to the fader when a touch-sensitive fader is used.

No fader update after release for motorized faders: When you move touch-sensitive faders, the fader messages to the controller are normally ignored by the controller. However, after release, the controller requires a positioning message. If this isn't sent, the fader will reset to its last position before being moved. Some controllers don't require this new position message. They jump to the new position from the old message to the new one first. To avoid this, deactivate sending of new positions.

The following options concern the **synchronization between controller range, mixer range, and the active track**. The controller range describes the section of available tracks in a project which can be controlled by the controller. The mixer range is the section of available tracks in the project that can be controlled by Samplitude's mixer.

Controller range follows active track: If this option is active, then the controller range is adjusted if the active track is changed so that the active track lies within the controller range.

Controller area follows controller range: If this option is active, a track within the controller range is activated when the controller range is changed. The track next to the previously activated track is also activated.

Mixer range follows controller range: If this option is active, changes to the controller range at the start of the mixer will be adjusted to the start of the controller range.

Mixer range follows controller range when active track is changed: This option is only sensible when "Controller range follows active track" is active. If a track outside of the controller range is active, then the controller range will be adjusted to it. These changes to the controller range readjust the mixer range.

Note: If you scroll through the mixer and simultaneously hold "Shift", the beginning of the controller range will be adjusted to the start of the mixer range.

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.

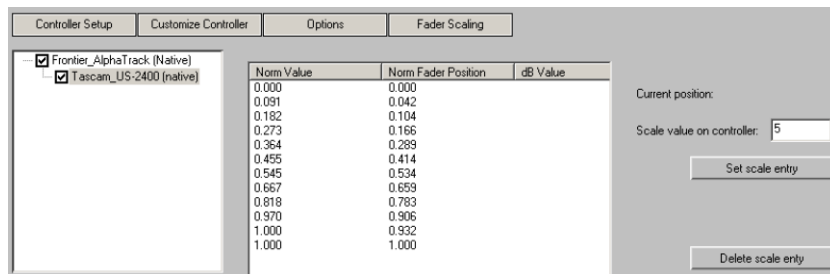
Control assignment display: Some controllers possess an additional display field for showing the current operational mode (track, EQ, AUX, etc.). If a controller does not such a display field, then deactivate control the display field so that control commands don't incorrectly influence other controller elements. This option updates Samplitude's automation modes optimally.

Control fader LEDs: Some controllers possess LED displays that are assigned to the dials. These can be used for visual display of the current parameter controlled by the dial. For controllers without LED displays, you can deactivate the display control.

Process messages with dialogs open: Normally, when hardware controller dialogs are open, MIDI messages are no longer processed to avoid unwanted changes. In some situations, adjusting or programming a controller requires that controller messages are able to be responded to. In this case, you can activate incoming messages while dialogs are open. It's a good idea to deactivate this option again once the controller is adjusted.

Hardware controller - fader scaling

Fader scaling offers you the option to adjust the position of the fader in the program to the pressed scale of the controller. Of course, fader control also functions without this adjustment. Control via a hardware controller can occur within a range of +6 dB to -100 dB. Normally, the fader position is right at the top for +6 dB and all the way down at -100 dB.



To set up new scaling, take the following steps:

1. Delete all previous entries. Click on the entry in the list and then press "**Delete scale entry**".
2. Enter the value "6" into the "**Scale entries on controller**" and move the fader all the way up (to the scale value +6 dB). Next, click "**Set scale entry**" to set the new entry.
3. Move the fader all the way down and enter the value "-100" into the "**Scale entries on controller**". Click "**Set scale entry**" again.
4. Insert additional scale positions (e.g. -10,-20, -40, etc.) in the same way.

To test if the scale matches, close the dialog with okay and compare the controller's fader positions with fader positions in the program mixer. Additional scale values may also be added later. To change a value, delete the previously learned value first.

Learn controller for plug-in/mixer elements

The respective active mode may be adjusted for individual settings. The **mixer's system menu** features the entry "**Learn hardware controller**" for this. Right click the icon in the title bar in the mixer to open this function.



Click the command "**Learn hardware controller**", activate a mixer element with the mouse, and then move the desired control element on the controller. The control element will now control the assigned mixer element. This assignment only changes the mode active at this time. This makes it possible to use controller elements that have been learned in the hardware controller dialog.

The **plug-ins** menus also include the entry "**Learn hardware controller**". The plug-in parameters may also be learned for a controller in the same way.



The changes made remain available until either "**Restore modes**" is clicked in the hardware controller dialog or until a change is made in the hardware controller dialog settings.

Here are the instructions for learning a controller:

1. Click the desired plug-in/mixer element or move it
2. Move the selected element on the hardware controller

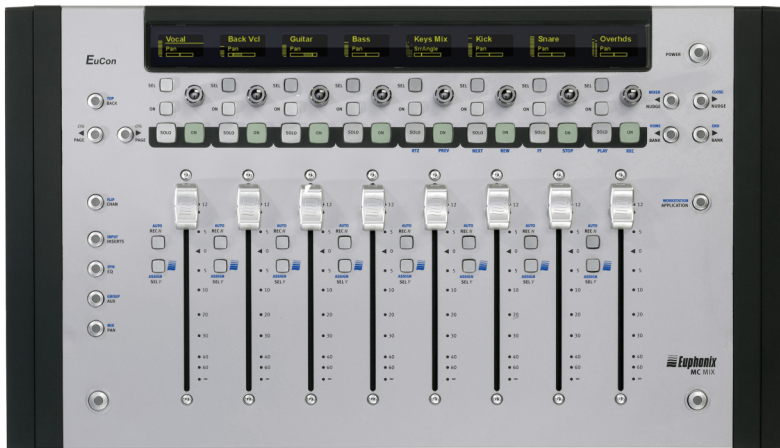
Pay attention to the following:

- The hardware controller element must have been learned in the hardware controller setup beforehand and may not be empty.
- Learned elements permanently modify the internal mode. Assignment is also available later for other projects.
- Assignment may be cleared again via the "**Restore modes**" button beside the internal mode.

Hardware controller presets

The following list features the most important presets for hardware controllers and integrating/configuring devices.

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.

Controller configuration

Track area:

Control element	Function	Shift function
SEL 1 ~ 8		
ON 1 ~ 8		
DIAL 1~8	Panorama	
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	Track arrange using EuControl

Navigation:

Control element	Function	Shift function
TOP BACK	Return from a subpage	Return to upper most page
< PAGE	Parameter page to the left e.g. in channel editing mode, AUX or Inserts	
PAGE >	Parameter page to the right e.g. in channel editing mode, AUX or Inserts	
FLIP CHAN	Channel editing mode (Phat-Channel)	Flip mode: KNOBS and FADERS swap functions
INPUT INSERTS	Plug-in mode	
DYN EQ	EQ mode	
GROUP AUX	AUX mode	
MIX PAN	Pan mode	
SHIFT	Shift key	
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

Observations:

The modes INPUT, DYN, GROUP and MIX are not available in the software.

Artist Control



1. Close your controller (according to manufacturer's instructions) and install the required driver software.
2. In case the files Sam.xml or Sequoia.xml don't exist in the driver software programs\Euphonics\EuCon\SQLR copy and paste it here from the folder program directory/Customize.
3. Go to system options/Hardware controller and select "activate EUCON".
4. Close the dialog. Your controller is now ready to use.

Controller configuration

Track area:

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	Track arrange 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 range:

Control element	Function	Shift function
DIAL 1~8	Function independent of view	
SOFT KEYS 1 ~ 12	Function independent of view	

Navigation:

Control element	Function	Shift function
MONITOR CONTROL ROOM	Monitor volume	Solo-Bus volume
TOP BACK	Return from a subpage	Return to upper most page
WORKSTATION APPLICATION	Switch the current Windows application	Workstation switch
< PAGE	Parameter page to the left e.g. in channel editing mode, AUX or Inserts	
PAGE >	Parameter page to the right e.g. in channel editing mode, AUX or Inserts	
MIXER < NUDGE	Controller track 1 to the left	Mixer
CLOSE NUDGE >	Controller track 1 to the right	Close application
HOME < BANK	Controller track 8 to the left	View goes to track 1
END BANK >	Controller track 8 to the right	View goes 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	

Observations:

The modes INPUT, DYN, GROUP and MIX are not available in the software.

Artist Transport

1. Close your controller (according to manufacturer's instructions) and install the required driver software.
2. In case the files Sam.xml or Sequoia.xml don't exist in the driver software programs\Euphonics\EuCon\SQLR copy and paste it here from the folder program directory/Customize.
3. Go to system options/Hardware controller and select "activate EUCON".
4. Close the dialog. Your controller is now ready to use.

Controller configuration

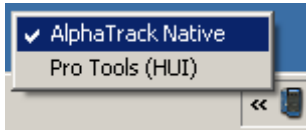
Copy Sam.xml or Sequoia.xml

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 key (0 ~ 9)	Number input	
-		
+		
.	Decimal place	
*	ESC/Cancel	
Enter	Enter	
TRANSPORT KEY 1	Cursor at project start	
TRANSPORT KEY 2	Fast rewind	
TRANSPORT KEY 3	Fast forward	
TRANSPORT KEY 4	Stop	
TRANSPORT KEY 5	Playback	
TRANSPORT KEY 6	Record	
TRANSPORT KEY 7	Loop mode	
WORKSTATION APPLICATION	Switch the current Windows application	Workstation switch
SHIFT	Shift key	

Frontier AlphaTrack



1. AlphaTrack is supported in "Native" mode. Right clicking the AlphaTrack symbol in the toolbar switches the device to AlphaTrack Native.



2. Go to Samplitude/Sequoia's system settings/hardware controller and select **AlphaTrack** using the **New** button.
3. Next, set the right side of the MIDI port to **AlphaTrack**.
4. Close the dialog. The controller is now operational.

Controller configuration

Control element	Function	SHIFT-Label	SHIFT - Function
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	Solo track		Global solo
MUTE	Mute track		Global mute
SHIFT	Shift		
PAN	Press first time: Pan mode Press second time: 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
< 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	Cursor at project start
>>	Fast forward	END	Cursor at project end
STOP	Stop	Esc	Cancel
PLAY	Play	Enter	Enter
RECORD	Record	Mode	
TOUCHPAD	Positioning		
Foot switch	Record		

Special modes

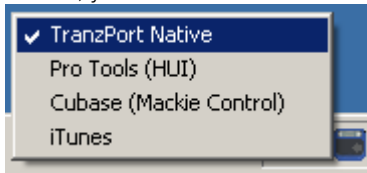
Mode	Control element	Rotate	Press
Pan mode	DIAL 1	Track selection	
	DIAL 2	To the marker	Set marker
	DIAL 3	PAN	Reset pan
ACTIVE CONTROL MODE	FADER (controls the active element in the mixer, track editor, or plug-in that must be selected with the mouse beforehand)		
	DIAL 1	PAN	
	DIAL 2	PAN	
	DIAL 3	PAN	
AUX mode	DIAL 1	Track selection	
	DIAL 2	AUX level	
	DIAL 3	AUX selection	
EQ mode	DIAL 1	Level	
	DIAL 2	Frequency	
	DIAL 3	quality	
	< TRACK	Previous EQ band	
	> TRACK	Next EQ band	
Plug-in mode Plug-in slot selection	DIAL 1	Track selection	
	DIAL 2		
	DIAL 3	Slot selection	
	< TRACK	Select plug-in	

	> TRACK	Select plug-in
Plug-in parameter	DIAL 1	Change parameters
	DIAL 2	Change parameters
	DIAL 3	Change parameters
	< TRACK	Previous parameters
	> TRACK	Next parameters

Frontier TranzPort



1. TranzPort is supported in "Native" mode. By right clicking the TranzPort symbol in the toolbar, you will switch the device to **TranzPort Native**.



2. Go to Samplitude/Sequoia's system settings/hardware controller and select the **Frontier TranzPort (Native)** entry from the list via the **New** button.
3. Next, set the right side of the MIDI port to **TranzPort**.
4. Close the dialog. The controller is now operational.

Controller configuration

Control element	Function	SHIFT - Function
< TRACK	Previous track	
> TRACK	Next track	
REC	Record track	Monitor track
MUTE	Mute track	Global mute
SOLO	Solo track	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	Cursor at project start
>>	Fast forward	Cursor at project end
STOP	Stop	Cancel
PLAY	Play	Enter
RECORD	Record	
DATA WHEEL	Positioning	Track volume
DATA WHEEL + STOP	Selected range	
Foot switch	Record	

JLCooper MCS-3800



1. Go to Samplitude/Sequoia's system settings/hardware controller and select the "JLCooper MCS-3800 (Native)" entry from the list using the "New" button.
2. Now set up the MIDI ports with the corresponding in and outputs of your MIDI interface on the right side.
3. Close the dialog. The controller is now operational.

Controller configuration

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	Solo track		
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
DIAL 1~5	Change parameters

Functions window

Control element	Function
F1	Punch marker mode
F2	Loop
F3	Metronome
F4	Synchronization
F5	
F6	
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
+/-	
LAST	
SET LOCATE	
MODE LOCATE	

Transport range

Control element	Function
REWIND	Fast rewind
FAST FORWARD	Fast forward
STOP	Stop
PLAY	Play
RECORD	Record

Special modes

Mode	Control element	Function
Track mode	DIAL 1	
	DIAL 2	
	DIAL 3	
	DIAL 4	Track pan
	DIAL 5	To the marker
AUX mode	DIAL 1	AUX level
	DIAL 2	AUX level
	DIAL 3	AUX level
	DIAL 4	AUX level
	DIAL 5	Selection AUX range
EQ mode	DIAL 1	Level EQ band 1
	DIAL 2	Frequency EQ band
	DIAL 3	Quality EQ band
	DIAL 4	
	DIAL 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)	

Notes

In "Shift" mode, the LED buttons will not refresh for the commands learned in "Shift" mode.

LEDs for Bank3 (time information) and Bank 4 (video window) are inactive.

Plug-in controls are inactive.

Logic Control

1. Go to Samplitude/Sequoia's system settings/hardware controller and select **Logic Control** from the list via the **New** button.
2. Now set up the MIDI ports with the corresponding in and outputs of the MIDI interface on the right side.
3. Close the dialog. The controller is now operational.

Controller configuration

TRACKS

Control element	Function	Modifier	Modifier function
DIAL 1~8	Pan; Eq; Aux (see special modes)		
REC/RDY 1~8	Activate track	SHIFT	Monitor track
SOLO 1~8	Solo track		
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: Invert functions of dials and faders
GLOBAL VIEW	Mixer

DISPLAY

Control element	Function
NAME/VALUE	
SMTP/BEATS	Timecode switch

FUNCTION BUTTONS

Control element	Function	Modifier	Modifier function
F1	Set / jump to marker 1	SHIFT CNTRL	Move marker 1 Delete marker 1
F2	Set / jump to marker 2	SHIFT CNTRL	Move marker 2 Delete marker 2
F3	Set / jump to marker 3	SHIFT CNTRL	Move marker 3 Delete marker 3
F4	Set / jump to marker 4	SHIFT CNTRL	Move marker 4 Delete marker 4
F5	Set / jump to marker 5	SHIFT CNTRL	Move marker 5 Delete marker 5
F6	Set / jump to marker 6	SHIFT CNTRL	Move marker 6 Delete marker 6
F7	Set / jump to marker 7	SHIFT CNTRL	Move marker 7 Delete marker 7
F8	Set / jump to marker 8	SHIFT CNTRL	Move marker 8 Delete marker 8

GLOBAL VIEW

Control element	Function
-----------------	----------

MIDI TRACKS

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		

TRANSPORT

Control element	Function	Modifier	Modifier function
MARKER	Marker mode (see table 2)		
NUDGE	Object mode (see table 2)		
CYCLE	Loop mode		
DROP	Punch marker mode		
REPLACE	Synchronization		
CLICK	Metronome		
SOLO	Activate LCD meter by holding "F1", "F2", "F3", or "F4"		
REWIND	Fast rewind	SHIFT	Cursor at project start
FAST FWD	Fast forward	SHIFT	Cursor at project end
STOP	Stop		
PLAY	Play		
RECORD	Record		
Cursor left	Range to left	ZOOM	Horizontal project zoom (-)
Cursor right	Clip to 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	Rotate	Press
Track mode	DIAL	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	DIAL	PAN	Reset
EQ mode	DIAL 1	Level EQ band 1	
	DIAL 2	Level EQ band 2	
	DIAL 3	Level EQ band 3	

	DIAL 4	Level EQ band 4	
	DIAL 5	Frequency EQ band 1	
	DIAL 6	Frequency EQ band 2	
	DIAL 7	Frequency EQ band 3	
	DIAL 8	Frequency EQ band 4	
AUX mode AUX 1~8 selectable using BANK	DIAL 1~8	AUX level	
AUX mode (phat channel) depending on selected track	DIAL 1	AUX 1 level	
	DIAL 2	AUX 2 level	
	DIAL 3	AUX 3 level	
	DIAL 4	AUX 4 level	
	DIAL 5	AUX 5 level	
	DIAL 6	AUX 6 level	
	DIAL 7		
	DIAL 8		
Plug-in mode	DIAL 1~8		Select plug-in
Plug-in parameter	DIAL 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

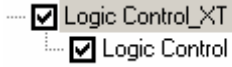
Note:

Crossfade editor (only Sequoia): Controller operation is inactive.

Logic Control XT

- Go to Samplitude/Sequoia's system settings/hardware controller and select **Logic Control XT** from the list via the **New** button.
- Position **Logic Control XT** under **Logic Control**
 - Logic Control
 - Logic Control_XT

If you have set up **Logic Control XT** to the left of **Logic Control**, the entries in the list will appear as follows



3. Now set up the MIDI ports with the corresponding in and outputs of the MIDI interface on the right side.
4. Close the dialog. The controller is now operational.

The functions correspond with the Logic Control.

Note: The number of tracks to be jumped when a bank is switched may be altered via the program's hardware controller setup options. These settings are available via **Logic Control**.

Mackie Control

1. Go to Samplitude/Sequoia's system settings/hardware controller and select "Mackie Control" from the list via the "**New**" button.
2. Now set up the MIDI ports with the corresponding in and outputs of the MIDI interface on the right side.
3. Close the dialog. The controller is now operational.

Controller configuration

TRACKS

Control element	Function	Modifier	Modifier function
DIAL 1~8	Pan; Eq; Aux (see special modes)		
REC/RDY 1~8	Activate track	SHIFT	Monitor track
SOLO 1~8	Solo track		
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: Invert functions of dials and faders
EDIT	Mixer

DISPLAY

Control element	Function
NAME/VALUE	
SMTP/BEATS	Timecode switch

FUNCTION BUTTONS

Control element	Function	Modifier	Modifier function
F1	Set / jump to marker 1	SHIFT CNTRL	Move marker 1 Delete marker 1
F2	Set / jump to marker 2	SHIFT CNTRL	Move marker 2 Delete marker 2
F3	Set / jump to marker 3	SHIFT CNTRL	Move marker 3 Delete marker 3
F4	Set / jump to marker 4	SHIFT CNTRL	Move marker 4 Delete marker 4
F5	Set / jump to marker 5	SHIFT CNTRL	Move marker 5 Delete marker 5
F6	Set / jump to marker 6	SHIFT CNTRL	Move marker 6 Delete marker 6
F7	Set / jump to marker 7	SHIFT CNTRL	Move marker 7 Delete marker 7
F8	Set / jump to marker 8	SHIFT CNTRL	Move marker 8 Delete marker 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		

FUNCTIONS

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	Cursor at project start
FAST FWD	Fast forward	SHIFT	Cursor at project end
STOP	Stop		
PLAY	Play		
RECORD	Record		
Cursor left	Range to left	ZOOM	Horizontal project zoom (-)
Cursor right	Clip to 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	Rotate	Press
Track mode	DIAL	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	DIAL	PAN	Reset
EQ mode	DIAL 1	Level EQ band 1	
	DIAL 2	Level EQ band 2	
	DIAL 3	Level EQ band 3	
	DIAL 4	Level EQ band 4	
	DIAL 5	Frequency EQ band 1	
	DIAL 6	Frequency EQ band 2	
	DIAL 7	Frequency EQ band 3	
	DIAL 8	Frequency EQ band 4	
AUX mode AUX 1~8 selectable using BANK	DIAL 1~8	AUX level	
AUX mode (phat channel) depending on selected track	DIAL 1	AUX 1 level	
	DIAL 2	AUX 2 level	
	DIAL 3	AUX 3 level	
	DIAL 4	AUX 4 level	
	DIAL 5	AUX 5 level	
	DIAL 6	AUX 6 level	
	DIAL 7		
	DIAL 8		
Plug-in mode	DIAL 1~8		Select plug-in
Plug-in parameter	DIAL 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

Note:

All statements refer to MCU firmware 2.1.2

Crossfade editor (only Sequoia): Controller operation is inactive.

Mackie Control XT

1. Go to Samplitude/Sequoia's system settings/hardware controller and select **Mackie Control XT** from the list via the **New** button.
2. Position **Mackie Control XT** under **Mackie Control**

```

-----  Mackie Control
         |
         |-----  Mackie Control_XT
  
```

If you have set up **Mackie Control XT** to the left of **Mackie Control**, the entries in the list will appear as follows

```

-----  Mackie Control_XT
         |
         |-----  Mackie Control
  
```

3. Now set up the MIDI ports with the corresponding in and outputs of the MIDI interface on the right side.
4. Close the dialog. The controller is now operational.

Functional assignment can be referenced in the chapter "Mackie Control".

Note: The number of tracks to be jumped when a bank switch is pressed may be changed in the program's hardware controller setup options. These settings are available in **Mackie Control**.

PreSonus FaderPort



1. Go to Samplitude/Sequoia's system settings/hardware controller and select the "PreSonus FaderPort (Native)" entry from the list using the **"New"** button.
2. Next, set the MIDI ports on the right side to USB audio device.
3. Close the dialog. Your controller is now ready to use.

Controller configuration

Control element	Function	SHIFT labeling	Value control element	SHIFT - Function	Return value for LED
FADER	Track volume				
PAN	Track pan				
MUTE	Mute track		Global mute	A01200	A01501
SOLO	Solo track		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)			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				A00C00	A00B01
TRNS	Transport console			A00D00	A00A01
UNDO	Undo	REDO	Restore	A00E00	A00901
SHIFT	Shift			A00200	A00501
PUNCH	Punch marker mode	PREV	Previous marker	A00100	A00601
USER	Metronome	NEXT	Next marker	A00000	A00701
LOOP	Loop mode	MARK	Insert marker	A00f00	A00801
REWIND	Fast rewind	START	Cursor at project start	A00300	A00401
FAST FORWARD	Fast forward	END	Cursor at project end	A00400	A00301
STOP	Stop			A00500	A00201
PLAY	Play			A00600	A00101

RECORD	Record	A00700	A00001
FOOT SWITCH	Record		

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)

Notes:

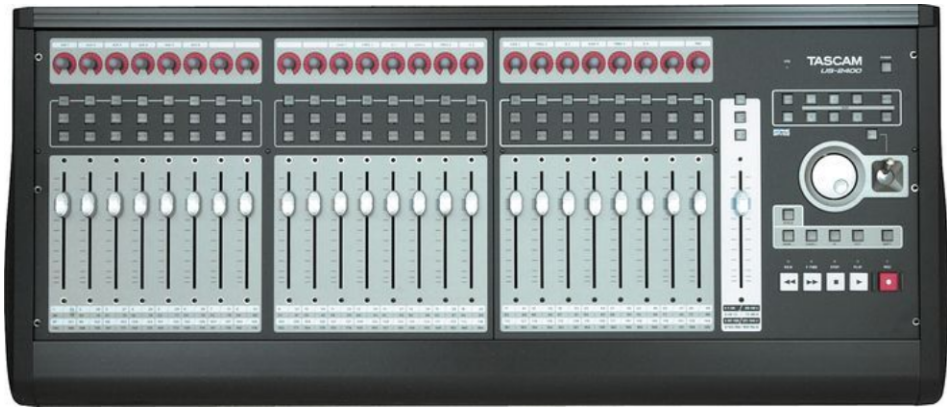
The foot switch is available for controller firmware 1.2 and up.

To learn control elements for the FaderPort, enter the response values to the controller for learning so that the button LEDs also function. Copy the value of the last column from table 1 of the corresponding control element into the "Send MIDI files" field. Here is an example of the correct arrangement for the MUTE keys.

Function Text	Tra..	Slot...	Par...	End..	<>	MIDI 1
HWC_TRANSPORT_WINDOW	1	1	-1	No		A00D00
HWC_UTIL_UNDO	1	1	-1	No		A00E01
PLAY	1	1	-1	No		A00600
REC	1	1	-1	No		A00700
REW	1	1	-1	No		A00300
SHIFT	1	1	-1	No		A00200
STOP	1	1	-1	No		A00500
TrackMonitoring	1	1	-1	No		A01000
TrackMute	1	1	-1	No		A01200
TrackRecRdy	1	1	-1	No		A01000
TrackSolo	1	1	-1	No		A01100
TrackVolumeFader	1	1	-1	No		B0005A
TrackVolumeFaderTouch	1	1	-1	No		A07F01
TrackVPot_Controller	1	1	-1	Yes		E0007E

Program function: Latest MIDI data: Send MIDI data:

Tascam US-2400



1. The Tascam US-2400 is supported in "Native" mode. Use the most current firmware. Start the device via the power button and hold down SEL (above the master fader) and CHAN. The device is now in "Native" mode.
2. Go to Samplitude/Sequoia's system settings/hardware controller and select the "Tascam US-2400 (Native)" entry from the list using the **"New"** button.
3. Next, set the MIDI ports on the right side to USB audio device.
4. Close the dialog. The controller is now operational.

Controller configuration

Track range

Control element	Function	Shift function
DIAL 1~24	Pan, EQ, AUX (see table 2)	
SEL 1~24	Select track	Track automation
SOLO 1~24	Solo track	Record track
MUTE 1~24	Mute track	Monitor track (ASIO only)
FADER 1~24 (with fadertouch)	Track volume	
SEL (master)		Copy automation mode to all tracks
CLR SOLO	Global solo	Global record
FLIP	Global mute	Flip mode: Invert functions of dials 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	DIAL 1~24 to AUX 1	Read automation
AUX 2	DIAL 1~24 to AUX 2	Touch automation
AUX 3	DIAL 1~24 to AUX 3	Latch automation
AUX 4	DIAL 1~24 to AUX 4	Overwrite automation
AUX 5	DIAL 1~24 to AUX 5	Trim automation
AUX 6	DIAL 1~24 to AUX 6	Automation off
METER	Activate level meter in dial display	
F-KEY	Active control mode (see special modes)	
NULL		
Shuttle wheel	Positioning	
Shuttle wheel + STOP	Select range	
SCRUB	Scrub mode	
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	Cursor at project start
FFW	Fast forward	Cursor at project end
STOP	Stop	Cancel
PLAY	Play	Enter
REC	Record	
Foot switch	Record	

Special modes

Mode	Control element	Function
Pan mode	DIAL 1~24	Panorama
EQ mode (1 press)	DIAL 1	Level EQ band 1
	DIAL 2	Frequency EQ band 1

	DIAL 3	Quality EQ band 1
	DIAL 4	
	DIAL 5	Level EQ band 2
	DIAL 6	Frequency EQ band 2
	DIAL 7	Quality EQ band 2
	DIAL 8	
	DIAL 9	Level EQ band 3
	DIAL 10	Frequency EQ band 3
	DIAL 11	Quality EQ band 3
	DIAL 12	
	DIAL 13	Level EQ band 4
	DIAL 14	Frequency EQ band 4
	DIAL 15	Quality EQ band 4
	DIAL 16	
EQ mode (2 press)	DIAL 1	Level EQ band 1
	DIAL 2	Level EQ band 2
	DIAL 3	Level EQ band 3
	DIAL 4	Level EQ band 4
	DIAL 5	Frequency EQ band 1
	DIAL 6	Frequency EQ band 2
	DIAL 7	Frequency EQ band 3
	DIAL 8	Frequency EQ band 4
	DIAL 9	Quality EQ band 1
	DIAL 10	Quality EQ band 2
	DIAL 11	Quality EQ band 3
	DIAL 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)	

Note:

The number of tracks to be jumped when a bank switch is pressed may be changed via the program's hardware controller setup options.

When using the TASCAM US-2400 as an exclusive recording panel, it's recommended that you relearn the SEL buttons as **RecordReady**.

Tascam FW-1884



1. Go to Samplitude/Sequoia's system settings/hardware controller and select the "Tascam FW-1884 (Mackie)" entry from the list using the **"New"** button.
2. Next, set the MIDI ports on the right side to "FW 1884 Control".
3. Close the dialog. The controller is now operational.

Controller configuration:

Encoder

Control element	Function	Shift function
FLIP	Flip mode: KNOBS and FADERS swap functions	
PAN	Pan mode	
AUX 1~6	AUX select	Record track
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	metronom		
LOOP/F5	"Loop" mode		
CUT	Split object	SHIFT	Separate object with alternative crossfade
DEL	Delete	CTRL	Delete with time/ripple
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		
FADER 1-8	Track volume		
FADER MASTER	Master volume	SHIFT	Trim automation
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		
REW	Fast rewind	SHIFT	Cursor to project start (Return to Zero)
FFW	Fast forward	SHIFT	Cursor 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

Notes

The following elements do not send MIDI values and may not be learned:

- PFL, HIGH, HI-MID, LOW-MID, LOW, GAIN, FREQ, Q, COMPUTER, MIDI CTRL, MON, CLOCK, ROUTE

- All elements with CTRL configuration may be learned in the control panel without providing a modifier, since the CTRL configuration already takes place in the controller and sends separate IDs.
- All elements with a CTRL configuration may 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.
- AUX elements LED doesn't light up correctly
- FLIP LED doesn't light up correctly
- The SoftLCD application that was installed with the FW-1884 drivers may be used to display track names and other Mackie Control-typical information

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 488)". 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

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
 Import audio... Ctrl + I
 Export WAV Ctrl + E

Project Properties >

Mixer setup... Ctrl + Shift + M
 Playback parameters P
 Project status Shift + I
 Track information... Alt + I
 Synchronization active G
 Synchronization setup... Shift + G
 Draw setup Shift + Tab
 System options Y

Edit menu

Undo
 Redo
 Cut into clip
 Delete
 Copy
 Insert clip
 Delete with time/ripple
 Clear
 Cut with time/ripple
 Copy + clear
 Copy as...
 Copy to clip store
 Overwrite with clip
 Paste from ClipStore
 Insert from clips with time/ripple
 Overwrite with clip

Ctrl + Z
 Ctrl + Y
 Ctrl + Shift + X
 Del
 C / Ctrl + C / Ctrl + Insert
 V / Shift + Insert
 Ctrl + Del
 Alt + Del
 Ctrl + Alt + X
 Ctrl + Alt + C
 Shift + C
 Ctrl + Shift + C
 Ctrl + Shift + X
 Ctrl + Shift + V
 Ctrl + Alt + V
 Alt + V / Insert

Range

Range all

A

Manipulate range

Move range start to left

Move range start to right

Move range end to left

Move range end to right

Fold range to left

Flip Range right

Beginning of Range -> 0

Beginning of Range <- 0

End of Range -> 0

End of Range <- 0

Range start to left marker

Range end to right marker

Split range

Note range

Other areas...

Get range

Get range length

Saver marker

Marker with name...

Markers with auto number

Place marker at recording position

Get markers

Recall last range

Field 1

Field 2

Field 3

Alt + Numeric Pad '/'

Alt + Numeric Pad '*'

Shift + Left arrow / Alt + Numeric Pad -

Shift + Right arrow / Alt + Numeric Pad '+'

Ctrl + Shift + Left arrow

Ctrl + Shift + Right arrow

Ctrl + Page up

Shift + Page up

Ctrl + Page down

Shift + Page down

Shift + F2

Shift + F3

B

Alt + f2 - F10 (exception: Alt + F4 and F9)

Alt + F11

Ctrl + F2 - F10

Ctrl + Shift + F2 - F10

Shift + 0-9

?

Shift+#

Alt + ?

1 - 9, 0

Shift + Back

Alt + Numeric Pad 1

Alt + Numeric pad 2

Alt + Numeric pad 3

Track menu

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
Edit Volume	Ctrl + Shift + K
Track EQ	Ctrl + Shift + F
Track information...	Alt + I
MIDI Controllers/VST Autom.	Ctrl + Alt + A
Track Freeze	Alt + Shift + F
Unfreeze Track	Alt + Shift + U
Activate next track	Alt + Arrow down
Activate previous track	Alt + Up arrow
Edit Pan	Ctrl + Shift + P

Object menu

Cutting objects

Duplicate and move	Ctrl + D
Separate with alternative crossfade	Shift + T
Splitting objects	T
Trim objects	Ctrl + T
Heal objects	Ctrl + Alt + T
Gluing objects	Ctrl + Alt + G

Move/edit objects/crossfade

Object step size 1

(Left) object to left	Ctrl + 1
Right object to left	Alt + 1
(Left) object to right	Ctrl + 2
Right object to right	Alt + 2
Move object(s) left	Ctrl + Alt + 1
Move object(s) right	Ctrl + Alt + 2
Object start to left	Ctrl + 3
Object start to right	Ctrl + 4
Object end to left	Alt + 3
Object end to right	Alt + 4
Move crossfade left	Ctrl + Alt + 3
Move crossfade right	Ctrl + Alt + 4
Object start offset to left	Ctrl + 5
Object start offset to right	Ctrl + 6
Object end offset to left	Alt + 5
Object end offset to right	Alt + 6
Increase left volume	Ctrl + 8
Reduce left volume	Ctrl + 7
Increase right volume	Alt + 8
Reduce right volume	Alt + 7
Increase volume	Ctrl + Alt + 8
Reduce volume	Ctrl + Alt + 7
(Left) object content to left	Ctrl + 9
(Left) object content to right	Ctrl + 0
Right object content to left	Alt + 9
Right object content to right	Alt + 0
Object(s) content to the left	Ctrl + Alt + 9
Move object(s) content right	Ctrl + Alt + 0

Object step size 2

(Left) object to left	Ctrl + Shift + 1
Right object to left	Alt + Shift + 1
(Left) object to right	Ctrl + Shift + 2
Right object to right	Alt + Shift + 2
Move object(s) left	Ctrl + Alt + Shift + 1
Move object(s) right	Ctrl + Alt + Shift + 2
Object start to left	Ctrl + Shift + 3
Object start to right	Ctrl + Shift + 4
Object end to left	Alt + Shift + 3

Object end to right	Alt + Shift + 4
Move crossfade left	Ctrl + Alt + Shift + 3
Move crossfade right	Ctrl + Alt + Shift + 4
Object start offset to left	Ctrl + Shift + 5
Object start offset to right	Ctrl + Shift + 6
Object end offset to left	Alt + Shift + 5
Object end offset to right	Alt + Shift + 6
Increase left volume	Ctrl + Shift + 8
Reduce left volume	Ctrl + Shift + 7
Increase right volume	Alt + Shift + 8
Reduce right volume	Alt + Shift + 7
Increase volume	Ctrl + Alt + Shift + 8
Reduce volume	Ctrl + Alt + Shift + 7
(Left) object content to left	Ctrl + Shift + 9
(Left) object content to right	Ctrl + Shift + 0
Right object content to left	Alt + Shift + 9
Right object content to right	Alt + Shift + 0
Object(s) content to the left	Ctrl + Alt + Shift + 9
Move object(s) content right	Ctrl + Alt + Shift + 0
Object to play cursor position	Ctrl + Alt + P
Object to original time position	Ctrl + Alt + O
Arranging objects	Ctrl + Alt + Shift + A
Mute objects	Ctrl + M
Creating looped objects	Ctrl + L

Select objects

Selecting all objects	Ctrl + A
Unselect all objects	Ctrl + Shift + A
Select next object	> / Ctrl + Alt + W
Select previous object	< / Ctrl + Alt + Q
Object lasso	Ctrl + Alt + L
Create group	Ctrl + G
Ungroup	Ctrl + U
Object name	Ctrl + N
Object editor	Ctrl + O
Object manager	Ctrl + Shift + O
Take manager	Ctrl + Alt + Shift + T

Play

Play / Play loop	Space
Play with preload	Shift + Space bar
Play with Preload	Shift + P
Only play selected objects	Ctrl + Space

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
Play upper audio material	Alt + Shift + Space
Play lower audio material	Ctrl + Alt + Spacebar
Stop and jump to current position	Num ,
Playback parameters	P

Playback mode

Auto scroll	Scroll
Soft autoscroll	Shift + Scroll
Scrubbing active	Alt + Shift + Cursor down
Scrub left	Alt + Shift + Cursor left
Scrub right	Alt + Shift + Cursor right
Record	R
Record options...	Shift + R
Input monitoring	Alt + Shift + M

Move play cursor

To beginning	Home
To end	End
Left move in page mode	Cursor left
Left move in scroll mode	Alt + Cursor left
Right move in page mode	Cursor right
Right move in scroll mode	Alt + Cursor right
Object border left	Ctrl + Q
Object edge right	Ctrl + W
Marker left	F2/Alt + Q
Marker right	F3/Alt + W

Real time effects

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

Managers

File Browser

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

Source list manager

Ctrl + Alt + Shift + S

Marker list input

Ctrl + Shift + L

Take Manager

Ctrl + Alt + Shift + T

Clips

A clip

Shift + B

Three clips

B

Activate next clip

Page down

Activate previous clip

Page up

Show 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 + Cursor left
Half section right	Ctrl + Alt + Cursor right
Clip to play cursor/last stop position	Ctrl + Alt + ,
Zoom in	Cursor up, Ctrl+ Cursor right
Reduce range	Cursor down, Ctrl+ Cursor left
Show all	Ctrl + Alt + Cursor up
Zoom to range	Ctrl + Alt + Cursor down

Vertical

Half section up	Shift + Cursor up
Half section down	Shift + Cursor down
Zoom into waveform	Ctrl + Cursor up
Zoom out of waveform	Ctrl + Cursor down

Window

Tile horizontal	Return (Enter)
Restore	Shift + Return
Mixer	M
Time display	Ctrl + Shift + Z
Visualization	Ctrl + Alt + Shift + V
Transport console	Ctrl + Shift + T
Close all windows	Ctrl + H

Help menu

Help	F1
Context help	Shift + F1

Mouse

Center click
Wheel

Start/stop play
Horizontal scrolling

Shift + wheel

Fine tune active
buttons/faders/wave zooming

+ Ctrl

Horizontal zooming

+ Alt

Vertical zooming

+ Ctrl + Alt

Horizontal and vertical zooming

+ Ctrl + Shift

Vertical scrolling in VIP (tracks)

Signal Flow

Record

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

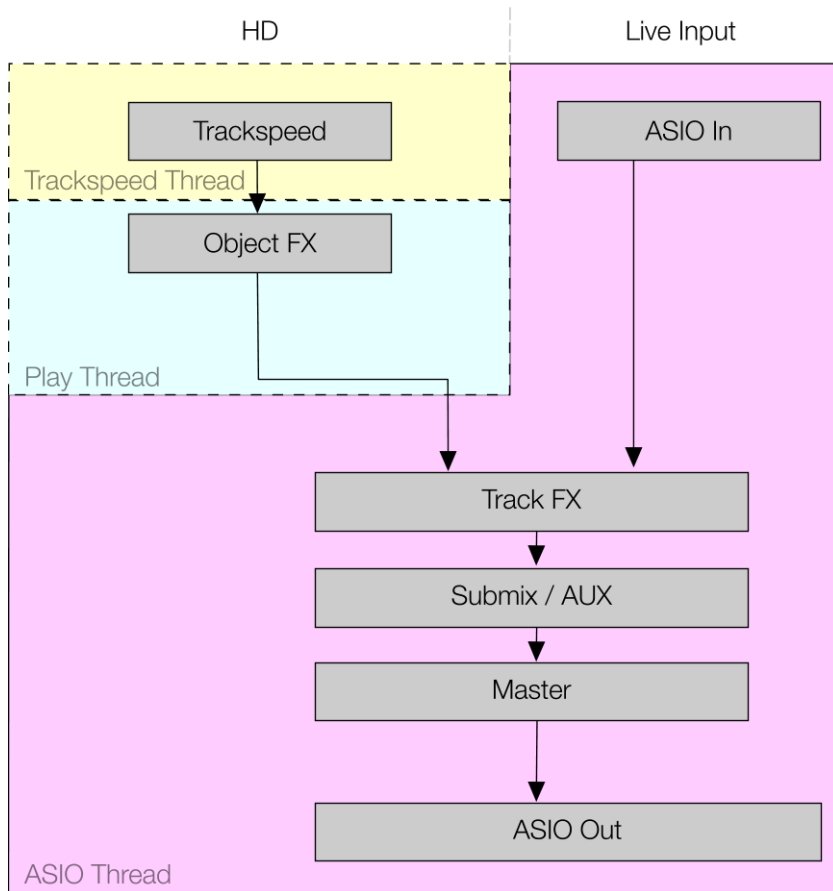
Playback

During playback, all files are read from the hard drive, and they pass through the track speed buffer and finally the effect sections. This not only ensures the allocation of effects, but also setting of the volume and panorama as well as buses and outputs. The signals are processed in packages at the size of the VIP buffer and are finally reproduced by the sound card via the ASIO buffer (ASIO only).

Signal flow with monitoring

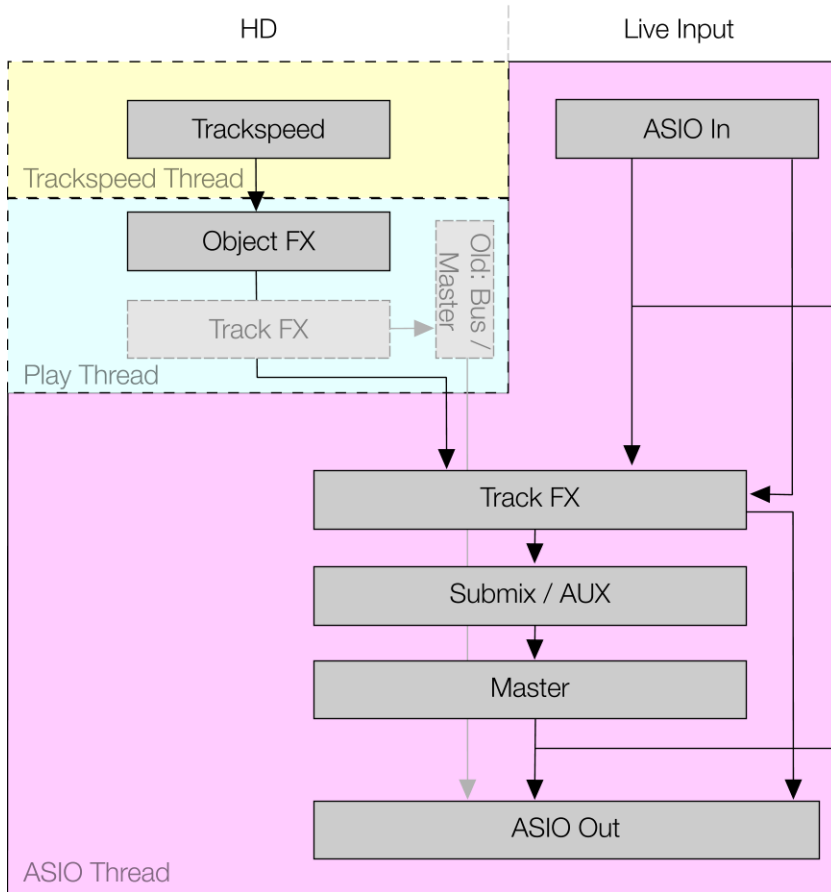
For all other models, three signal flows occur simultaneously: playback of the recorded tracks, recording of tracks to be recorded, and monitoring of the signals of the input.

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



Hardware monitoring

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



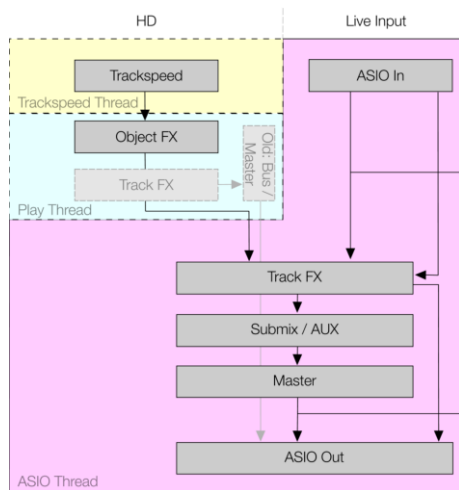
Software monitoring

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

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

For more information about the Hybrid Engine, please also read the section "System options -> Monitoring settings (view page 27)".

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



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 811)"

Edit menu "Range" > Split Range (view page 522)"

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

Clip for audio files (Wave projects)

The clip is a buffer 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 a wave project. 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 > 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.

VirtClip for virtual projects

While the clip is responsible for wave projects, the data from virtual projects (i.e. objects in the virtual project) is buffered in the VirtClip (virtual clip). The clip window is normally hidden; however, you can make it visible with the command "View > > Window > Iconize all Wave projects".

VirtClips are used as a location to store objects that were cut or copied out of a VIP window. Technically speaking the VirtClip is a special VIP window. It can therefore be edited like a virtual project. The content of the VirtClip can also be dragged onto any VIP window.

The size and track numbers of the VirtClip correspond with the range copied or cut from the original VIP window. If this is only a single object, the VirtClip will only consist of one track with a single object. However, if a range spanning over five tracks is cut out or copied, the VirtClip also contains five tracks with several objects.

You can easily exchange audio material between two windows with the help of the VirtClip.

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).

G

Grid/marker bar

The grid/marker bar is located above the first visible track. This is divided into two halves.

Marker bar/upper half: Markers may be placed in this section.

Left mouse click: Sets the play cursor.

Right mouse click: Accesses the context menu with entries for setting/getting the marker, etc.

Grid bar/lower half: The grid bar displays the project time in relation to the selected measurement unit. You can also select ranges if this option is not directly available in the tracks via the selected mouse mode (e.g. "Object/curve" mode).

Left mouse click: Sets the play cursor.

Left mouse click and drag: Opens a new range.

Right mouse click: Context menu including grid and raster entries.

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 when 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, this latency is disruptive even at 5 ms.

Response latency is the delay between activation of a sound producing or sound modifying control element and the audible change or generation of the sound. The response latency is made up of the sum of the output latency of the sound card, the latency of used effects, the latency of the play engine, and possibly the latency of the MIDI input/graphical user interface latency of the PC operating system.

When using MIDI-controlled synthesizers (VSTi's), the response latency is similarly critical, much like audio latency, in 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 caused by the effect in the lower control element bar. The latency of the sound card in conjunction with the driver being used is displayed in "System/audio options ("Y"). The overall 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.

Level fader

Volume control

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" 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.

Menu

Menus are located in the Samplitude main window, directly below the title bar. Each of these menu entries may be assigned a keyboard shortcut.

They may also appear in other program parts or dialogs.

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 being recorded is played back simultaneously via the output devices. Different modes are available depending on the driver model used and the monitoring mode. Monitoring has nothing to do with playback of the recorded signal. Only the signal at the input is involved.

Mouse mode

Mouse modes add specially adapted functions to the mouse.

O**Object editor**

The object editor enables you to edit the properties of any audio object exactly. For example, the length of the object may be entered numerically or the fade characteristics may be determined here.

Real-time effects, EQ, and dynamics editing may be applied to the object without having to access track effects in the mixer. The selected processing does not depend on any following changes to the object position.

Object-oriented

Object-oriented work describes a method that makes it possible to apply various changes to the audio material connected with the object virtually, i.e. without affecting the actual audio material (data on the hard drive). 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

Re-route or allocate internally

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.

Setup button in the VIP

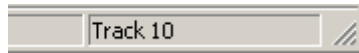
The 8 buttons at the lower left edge of a virtual project may store 4 setups (S1...S4, incl. zoom level, display position, and display mode) and 4 zoom levels (Z1...Z4).

To save, click the button while holding the "Shift" key. This allows you to quickly jump between certain positions and zoom levels without manually zooming in and out.

Right clicking a button opens a context menu containing options for loading.

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 via the "View > Toolbars > Status bar" menu item.

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 AUX bus

AUX buses are usually used to apply send effects. A Surround AUX bus has the same function, but with the option to integrate Surround effects. It may be sent from a normal track (see stereo and mono signal processing in Surround projects) or from a Surround bus. The AUX bus itself may also feed a Surround bus (if the track number is higher).

Surround sound in Samplitude

Surround master

From a technical viewpoint, the Surround master is a Surround bus of a Surround project that has the additional property of assigning the audio devices, but cannot feed a Surround bus itself.

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 panorama module 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.

Toolbar

Toolbars contain the tools that apply certain commands by clicking. These are located above and below the project window.

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. Tracks 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. This contains various functions including mixer elements, buttons for loading plug-ins, and other track parameters.



Track header with activated mute



V

VIP - virtual projects

Virtual projects (VIP) are the top level of audio editing in Samplitude. In virtual projects, objects may be 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 refers to the audio data.

A virtual project has two display modes. Use the "Tab" key to toggle between the two modes. The shortcut "Shift + Tab" may be used to open the editor for defining display properties.

VirtClip

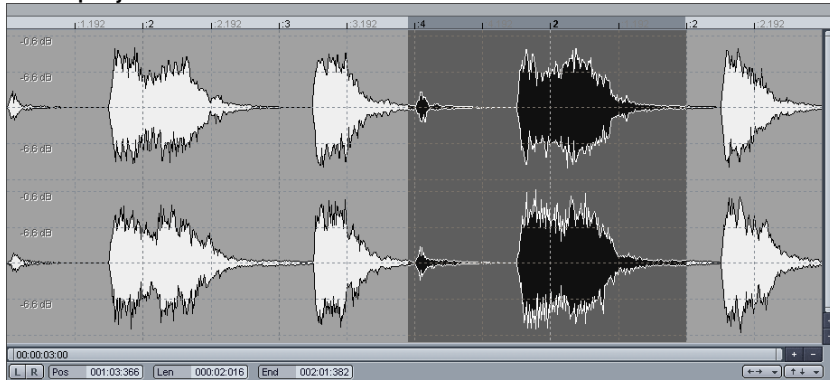
see "Clip (view page 426)"

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 108)".

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.

Z

Zoom

Use the zoom functions to display a larger or smaller part of the virtual project in the window. At higher zoom levels, the display is more detailed. Automation curves can be edited more easily when a higher zoom level is selected.

To view larger projects at their full length, you have to zoom out of the VIP window, making the display less detailed, but displaying more content.

Features a series of functions for working with various zoom levels Samplitude.

Menu reference

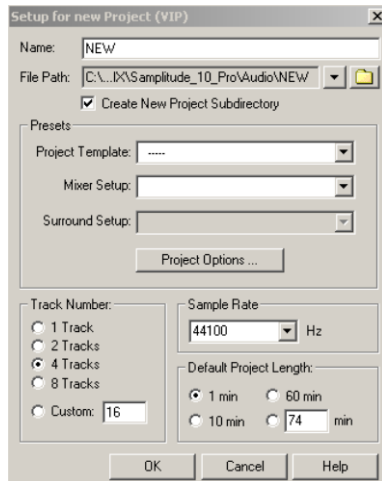
Extensive information on the individual menu points of Samplitude can be found in the following sections.

The menus in Samplitude may also be individually customized, i. e. menu items may be hidden or shown. If menu items are described in the following sections that are not present in your menus, then they are probably only hidden. To display them again, activate them in the menu settings via "File -> Program settings > Edit keyboard shortcuts and menus (view page 488)".

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 449)"

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 33)" 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 531)" 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 Project

You may open various file types here and load them into Samplitude. To load audio files, use the command "Load/import -> Load audio file".

Virtual Project (*.VIP)

Use this menu command to load a multi-track 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.

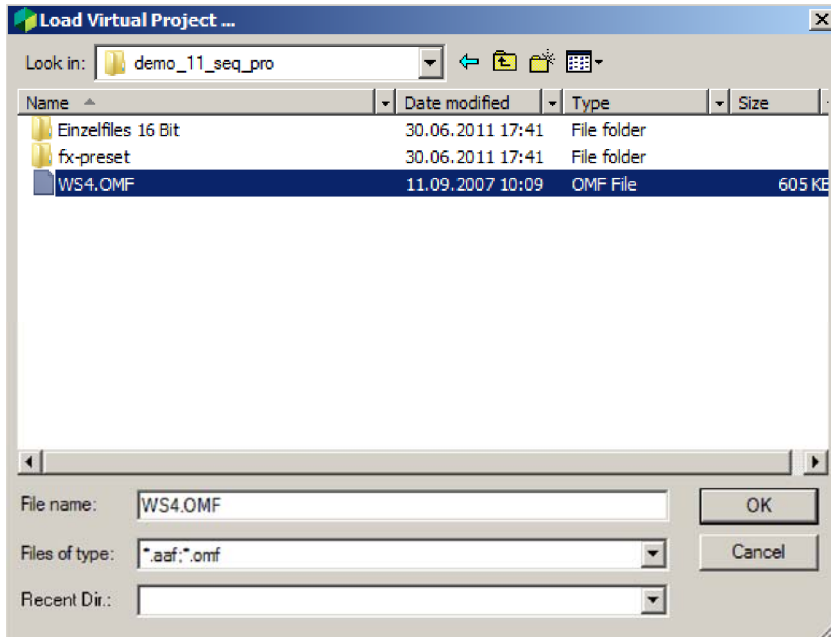
HD Wave (*.HDP)

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 792)).

Import AAF/OMF



The **AAF import** transfers the following content:

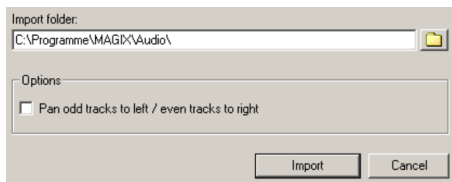
- Object position/wave offset - sample exact
- Object fade in/out - linear only
- Object crossfade - linear only
- Object volume
- Object pan
- Object volume automation
- Object pan automation
- Track names
- Track Volume
- Track Pan
- Track Volume Automation
- Track Pan Automation
- VIP markers
- Timecode 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.



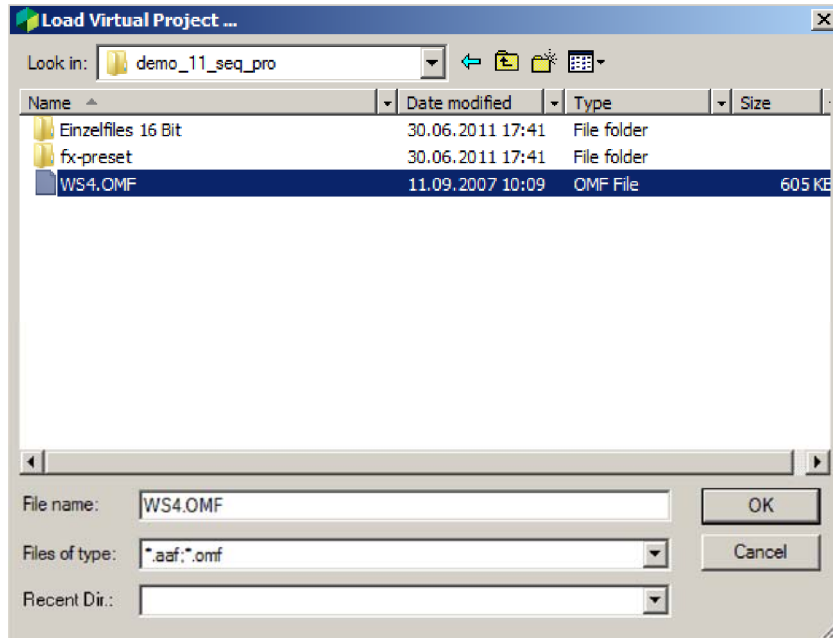
Object (*.OBJ)

An object contains 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.

Edit list (*.aaf, *.omf) – optional



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.

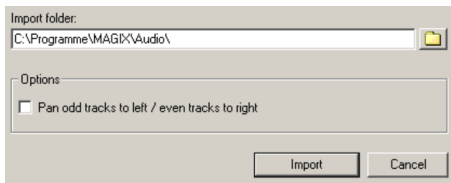


Table of Content (*.TOC)

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.

Load/Import

Load Audio File

This opens a dialog for loading 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.

Several files may also be loaded 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 that refer to loaded audio files are created simultaneously. 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 sec. after the last object. The length of the pause may be adjusted in the "CD/DVD" menu via "Set pause index".

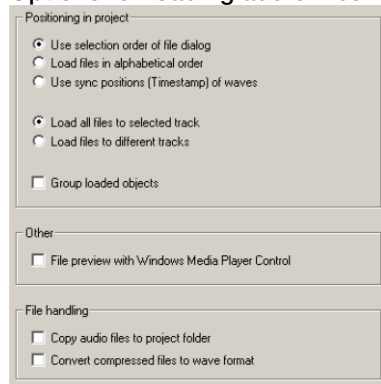
Each audio file may be previewed before it is loaded.

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 preview function again at any time.

The button at the bottom right of the dialog opens the options for loading audio files (view page 445).

Options for loading audio files



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: With this option activated, Samplitude sorts the selected files alphabetically in the VIP.

Use sync positions: 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.

Group loaded objects: All loaded files are grouped. They can be ungrouped anytime.

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 folder: Files are copied into the corresponding project folder automatically.

Convert compressed files to wave format: Compressed audio formats such as MP3 may also be loaded and played directly by Samplitude. This requires increased CPU calculation. Activate this option to convert files like this into an uncompressed wave format.

Import Audio

This menu item is hidden by default. To display it again, activate it in the menu settings via "File -> Program Preferences > Edit Keyboard Shortcuts and Menu (view page 488)".

This lets you import all supported file formats explicitly as Wave Projects.

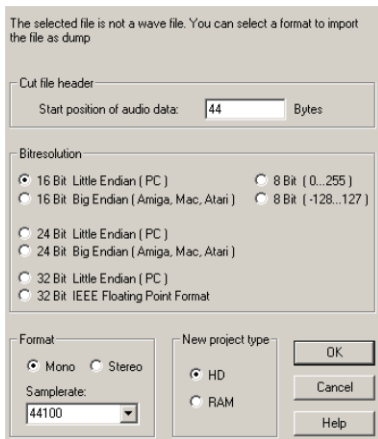
This is only necessary for importing file formats which can't be loaded by Samplitude using "Load audio file".

In contrast to the "Load audio file" function, when importing audio files wave formats are converted and copied to the hard drive. You can also use the command if you want to work in a copy of the file within the project.

Shortcut: Ctrl + I

Hint: You can increase the performance of your computer if you import compressed formats such as MP3 or WMA instead of loading them. It helps to avoid real time decompression

Use the "**Import raw data (dump)**" to try importing 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.



Import as Dump

Imports a wave or MP3 project as raw audio data (PCM) in RAW format (Little Endian for PC, Big Endian for Amiga, Mac, Atari).

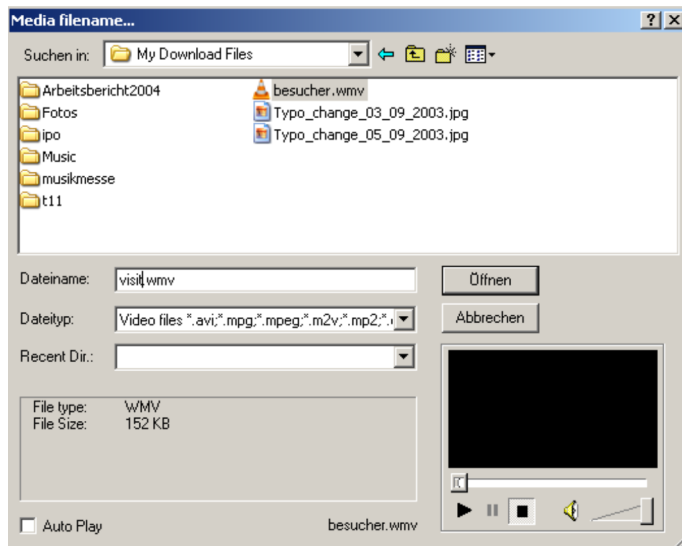
Load MIDI File

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 242)".

Keyboard shortcut: Shift + M

Load Video File

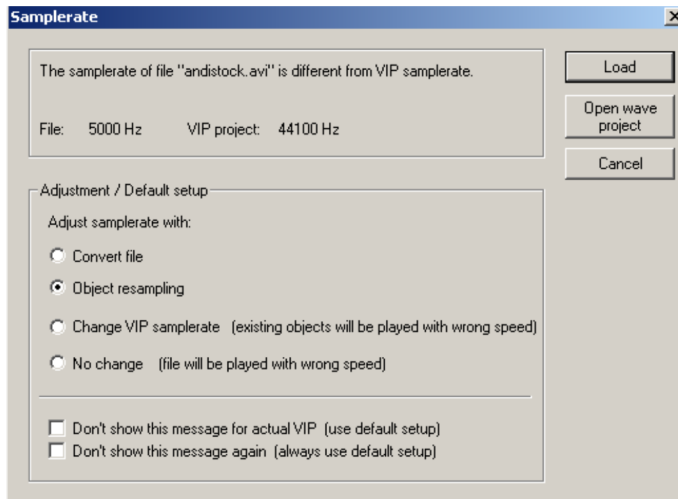
This dialog helps load a video into the current project. After selecting a video, preview it or inspect its properties. If a check is placed next to "**auto play**", the selected video file will play automatically in the preview window.



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 483)".
- Adjust VIP sample rate (existing objects are played back at incorrect speed)

- No change (file plays back at incorrect speed)



Detailed information about displaying video objects is available in "Video setup (view page 358)".

Import Audio CD Track(s)

This command imports the audio contents from a DVD into Samplitude (view page 777).

Import Audio DVD

This command imports the audio contents of a DVD into Samplitude (view page 780).

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 multi-track 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 Audio

This menu point allows you to export projects in various formats.

The export dialogs for each format are almost identical.

1. **Selection of path and file name.**
2. **Format settings:** This dialog features selection and settings options for the corresponding export format.

3. **Export settings:** Specify which area of the project should be exported here. Multiple files may be exported, provided their starting points have been specified with the placement of track markers or other markers. This way, an almost automatic export of individual files from within an arrangement is possible.

Note: The audio data from HD wave projects may be further processed or imported directly as wave files from HD wave projects. The use of the "Export" function is only necessary when the data is to be copied.

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 498)".

MP3...

Exports the project as MPEG Layer 3 with the included encoder. By clicking on the format settings, a new window opens where encoder options can be defined. Set output format, encoder quality, padding mode, stereo coding, content options, VBR (variable bit rate) options, and MP3 ID editor labeling.

MP3 with external encoder

This menu item is hidden by default. To display it again, activate it in the menu settings via "File -> Program Preferences > Edit Keyboard Shortcuts and Menu (view page 488)".

Here you can choose an external Encoder, which your project will export as MPEG Layer 3. Here, the format parameters, the audio stream as well as the new file name for the exported MP3 are transferred.

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.

Real Audio...

Exports the project in Real Audio format (.RA).

Real Audio is an audio format designed especially for Internet applications. This is especially useful for streaming audio. In format the settings, specify the bit rate of the transmission speed (modem, ISDN, etc.). Similarly, details on the content and copyright information may also be added.

Content settings: Select presets for the type of audio or video material (only voice/background music, music, stereo music) here.

Clip information: Enter author, video name, etc., which appear in the Real Player during playback here.

Multi-rate file (Real Server): Streaming audio requires a server application that must be installed on the server and from which the audio for playback ("streamed") in real time. If you don't have this application, it will have to be downloaded. "Real Server" is a server application that enables streaming audio in Real Audio format.

If the "Multi-rate file" option is activated, a multi-rate file can be created that includes several files in various bandwidths. Multi-rate files allow the selection of multiple target groups.

AAC...

Exports the project to an AAC file. You can also set the encoder settings in the advanced dialog, which is opened via the "Format settings" button. You can also set the quality and transport format (MPEG 4 or ADTS) here. AAC export uses the file extension *.m4a.

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.

AIFF File with QuickTime...

Exports the project in AIFF format. In this case, QuickTime is used as the encoder. This format also enables streaming playback of audio and video files over the Internet.

Like with Real Media exports, the corresponding video size, frame rate, and codec settings may be adjusted for "AIFF with QuickTime" export as well. This function requires the installation of QuickTime on your computer.

The following export options are available: Format (linear PCM, A-Law 2:1, IMA 4:1, MACE 3:1, MACE 6:1, QDesign Music 2, Qualcomm PureVoice™, μ -Law 2:1), channels (mono, stereo, quadrophonic, 5.0, 5.1), sample rate, sampling rate conversion quality as well as linear PCM settings in 8, 16, 24, 32 and 64-bit file format.

Mono / Stereo Convert

Stereo Wave -> 2 Mono

This function saves both the left and right channels of a stereo wave file (file.wav) separately as a mono file (file_left.wav, file_right.wav).

"Stereo to 2 Mono" may also be applied to objects in the VIP. Stereo objects to be split are replaced by two mono objects that are arranged one below the other and which feature panorama settings at the object level to the left or right.

2 Mono -> Stereo Wave

This combines two mono wave projects to one stereo wave project.

After executing this command, a dialog opens displaying all loaded mono files. You can load additional files with "Load file". From the opened wave projects, select the right and left files with the "^" keys. With "<->" keys you can switch channels. After clicking "Link files", specify the name of the new file to be created in the project folder.

Please note that only mono wave projects of the same word width and sample rate can be linked. The length of both projects are adapted to one another, the longer mono file sets the total length.

LR Wave -> 2 Mono

LR wave projects may be split into two independent mono projects using this function.

LR Wave -> 1 Mono

The active LR wave project is set to mono mode. In this case, the two channels are mixed together by adding the corresponding samples each at 100% and then splitting the sum in two in order to avoid overmodulation. This corresponds to sinking the level by 6 dB.

2 Mono -> LR Wave

This creates an LR wave project with two channels from two mono wave projects, in which case one mono wave project in panorama view is positioned to the left, and the other one to the right. Select both mono files with consecutive mouse clicks. In the following dialog you can decide to "Link". "Convert to stereo" creates a stereo wave file.

1 Mono -> LR Wave

In this case, LR wave project with two identical channels is created from a mono wave project. The original sample is duplicated here.

Save in format

Use this function to save wave projects in various formats.

This can be useful when, for example, RAM projects are to be converted into HD Wave Projects or LR Wave Projects (two mono samples linked to one another) are to be converted into Stereo Wave Projects.

In "**Area settings**", select whether only the **selected area** or the **entire wave project** should be saved in the new format.

Under "**General options**", select whether the **maximum amplitude** should be displayed or not and if the **master effects** should also be copied when the file is saved.

The **Dithering Options** (view page 498) 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 ratio) 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 as Dump

This menu item is hidden by default. To display it again, activate it in the menu settings via "File> Program Preferences > Edit Keyboard Shortcuts and Menu (view page 488)".

Export a Wave or MP3 Project as raw audio data (PCM) in RAW format (Little Endian for PC, Big Endian for Amiga, Mac, Atari).

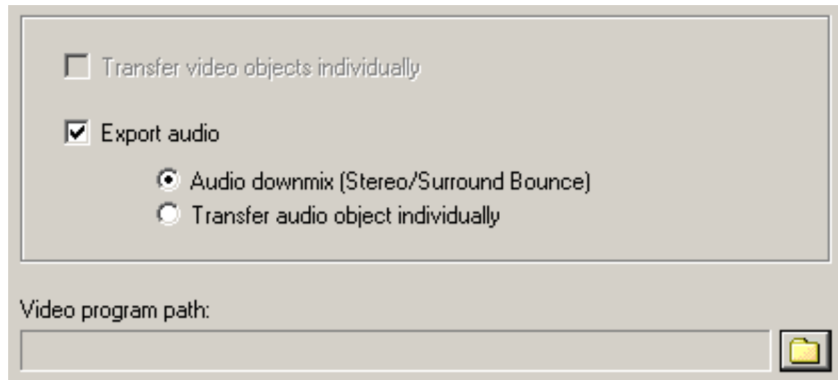
32-bit float as 16-bit / 20-bit / 24-bit

This menu item is hidden by default. To display it again, activate it in the menu settings via "File> Program Preferences > Edit Keyboard Shortcuts and Menu (view page 488)".

Exports the (32-bit float) project as a wave file with 16 / 20 / 24-bit. This takes into account the Dithering options (view page 498).

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.

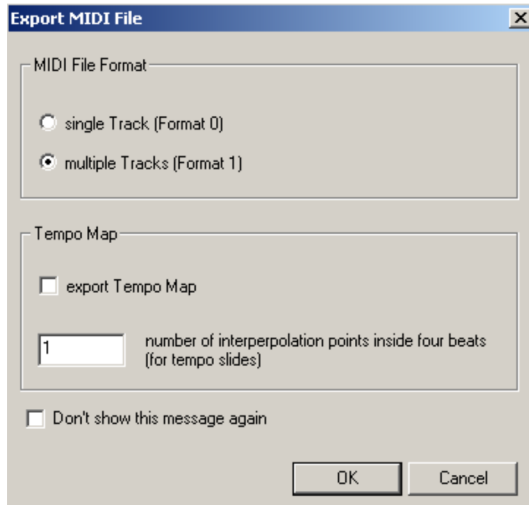
Set the source file under "Video source".

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.

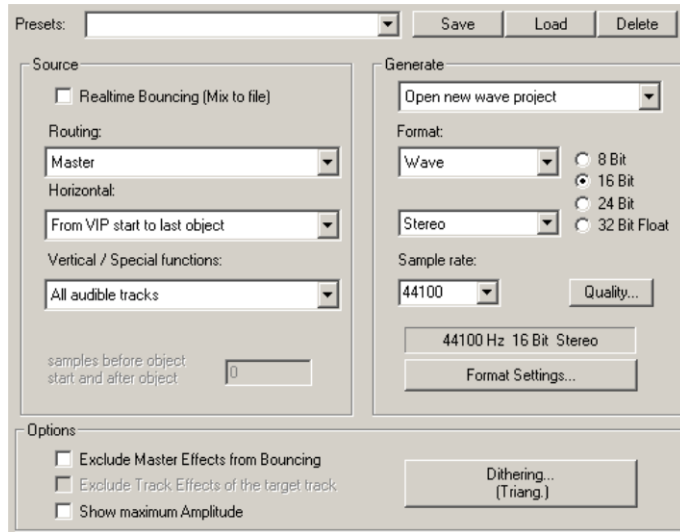
List Export

A current **marker list**, **object list**, or **track list** for your project may be written into text file and stored in the project folder.

Trackbouncing

This command can be applied to compile a project, a sought range or selected objects, and tracks of a virtual multi-track project (VIP) into one audio file and save them under one name. Here, all realtime editing (cuts, crossfades, all mixer settings including mixer effects and DirectX plug-ins, volume and panorama curves, realtime effects in the Object Editor, etc.) can be integrated into the new project.

If you bounce into the 32-bit float format, overmodulation cannot occur.



Normalize your project before bouncing using the master normalization function in the Mixer ("N" key) in order to get an precisely modulated file. You can always save the affected settings as presets.

Trackbouncing settings: Source

Routing:

In this field you determine the outputs to be written to the file.

Master: The mixer's master output is written as a file. This is the standard for all exporting processes.

All outputs (Ignore device routing): All tracks that have not been muted are calculated together, independent of the routing involved. Suitable for bouncing at object level.

Surround master (view page 240): This bounces the surround mix.

Surround + Stereo Master: The surround mix is bounced together with the stereo mix.

All outputs (Use device routing): All tracks that have not been muted are calculated together to the individual outputs according to their routing.

Horizontal:

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.

Vertical/Special:

All audible tracks: All tracks that have not been muted are calculated into a new file.

Selected tracks only: All of the selected tracks are bounced together.

All audible tracks individually (Multi-track bounce): All tracks that have not been muted are individually calculated into a new file.

Bounce all selected objects individually: All selected objects are bounced individually. In this case, all object effects are used. 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: All selected objects on a track are processed and written into a new file including object effects.

Note: The two previously mentioned options correspond to the "Object freeze" function / the "Glue objects" function in the "Object" menu. The difference that the file names may be specified individually and that an "Unfreeze" function is not available.

Samples before object start and after object end: This function allows additional samples to be included in the file to be calculated.

Note: If the selected track is a submix bus, all tracks that are routed to this bus are also bounced. This also works recursively, i.e. if these tracks are busses. This permits fast mixing of individual groups.

Track Bouncing Settings: Generate

Create new wave file: The selected range or the virtual project (VIP) creates a new wave file.

Open new wave project: The selected range/virtual project (VIP) is converted into a wave project.

Create new object in VIP: The bounced objects are inserted as new objects in a new track in the existing VIP.

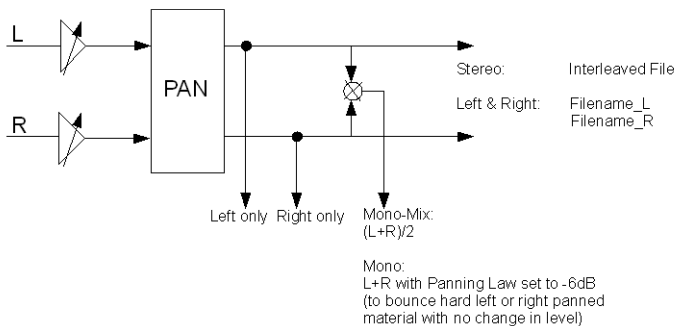
Replace objects in VIP: The objects used in trackbouncing are removed and the result is inserted into the selected track.

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 settings: 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 may 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**" may 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 240) in "Source -> Routing", then the formats "Mono", "Stereo", and "Interleaved file (RIFF 64)" will be available for selection.

Note: All files created as a result of trackbouncing are opened in Samplitude.

You cannot bounce to the same project twice in succession. You 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 "File -> Export" (view page 449) command instead of the trackbouncing dialog.

All formats that cannot be directly opened by Samplitude cannot be created in the trackbouncing window. For bouncing into Real or WMA formats, use the "File -> Export (view page 449)" command.

Trackbouncing criteria: Options

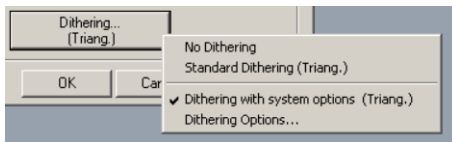
Exclude master effects from bouncing: Master effects are not included during bouncing.

Exclude track effects of the target track: Track effects of the target track are not included if they are not identical with a source track.

These options are important in combination with the "Create" functions "New separate object in current VIP" and "Replace objects in current VIP". Note: these are set automatically so that the track effects are not applied twice.

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 498)".

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.

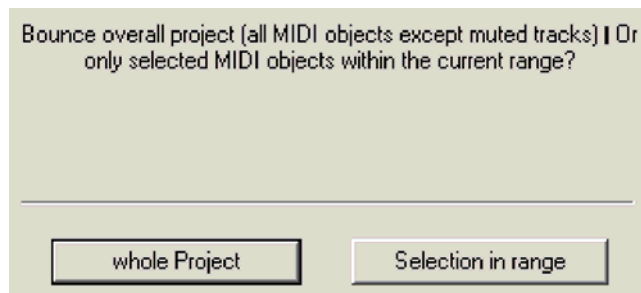
If the rate of your objects is higher than 16-bits, a dialog box will ask you whether the new file will be created as a 32-bit float or 16-bit file:

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.

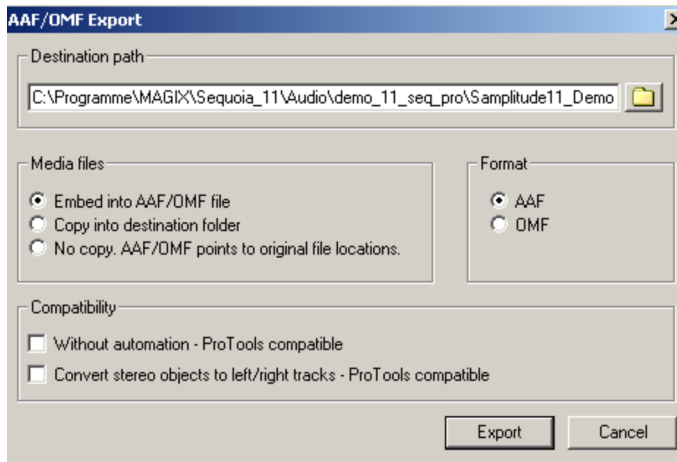
MIDI bouncing

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 563)" function, which replaces the MIDI objects with audio objects in the software instrument's audio return signal.

Export project as AAF/OMF



The **AAF Export** transfers the following content:

- Object position/wave offset - sample exact
- Object fade in/out - linear only
- Object crossfade - linear only
- Object volume
- Object pan
- Object volume automation
- Object pan automation
- Track names
- Track Volume
- Track Pan
- Track Volume Automation
- Track Pan Automation
- VIP markers
- Timecode offset
- Timecode format (24, 25, 30 fps)
- Processing timestamps in audio files

The **OMF Export** transfers the following content:

- Object position/wave offset - sample exact
- Object fade in/out - linear only
- Object crossfade - linear only
- Object volume
- Track names

Note: During AAF/OMF export, media files will always be converted to WAV files. You can't export video files. Corresponding objects will not be transferred; the video track therefore remains empty.

Destination path

Enter here where you would like to export the AAF/OMF file.

Media files

- **Embed in AAF/OMF file:** If you select this option when exporting then the whole project will be saved as a single file, which contains both media content and metadata.
- **Copy into target folder:** If you select this option, media files will be placed next to the AAF/OMF into the target folder.
- **Don't copy. AAF/OMF refers to original paths:** If you select this option, no media files will be exported – the AAF/OMF file will only refer to the original media file paths.

Note: Select the "Embed in AAF/OMF file" or "Copy to destination folder" if you want to have the entire project together with the corresponding Audio/Video files as AAF/OMF files in another studio.

Format

Select whether you want to export the file in AAF format or in OMF format.

Compatibility

Without automation curves (Pro Tools-compatible): If you select this option, AAF/OMF export will take place in a Pro Tools-compatible fashion. Track volume and panorama as well as automation curves will not be transferred.

Convert stereo objects into left/right tracks automatically - Pro Tools compatible: Stereo objects are not supported by Pro Tools. Convert the corresponding objects into Pro Tools-compatible objects prior to AAF/OMF export as required.

Note: Timestamps saved with ProTools in Frames for improved compatibility.

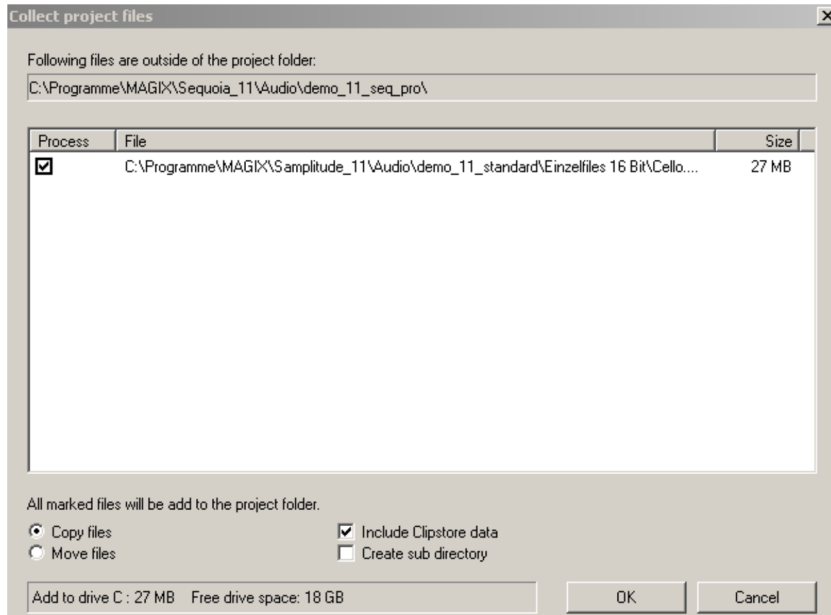
Rename Project...

The current project may be renamed using this dialog. For HD wave projects, all the files involved will be correspondingly renamed.

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 either 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 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 new 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.

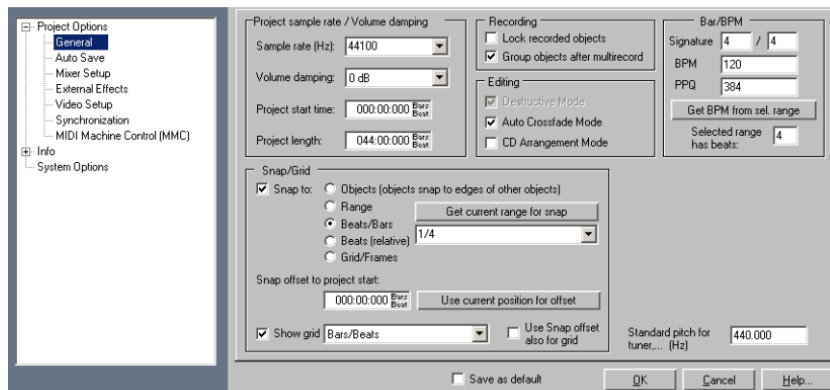
Project Properties

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 562) ("Object -> Set hotspot/delete hotspot (view page 562)"). The reference point is normally the front edge of the object that is to be moved. As soon as a hotspot has been defined for the object, this is what will be used. If several objects are selected, alignment takes place at the forward edge of the hotspot of the object that was selected last, and is currently located under the mouse cursor. The intervals between the selected projects stay constant.



Grid: Checking the tick box turns snap on globally.

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

Snap to range: Activates the range grid with the option to use the current range as the basis for snapping.

Grid/bars: Activates a grid with bars as the basis. The objects snap to the the next beat grid.

Grid/bars (Relative): Activates a grid with bars as the basis. This means that the relative distances between the objects adhere to the beat grid when you move them about.

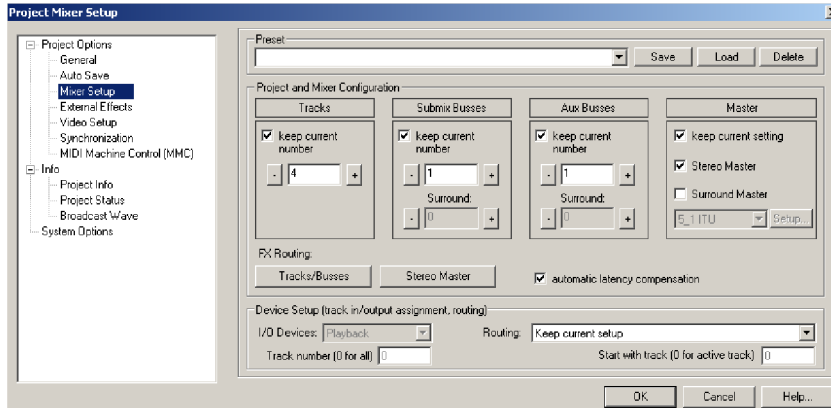
Grid/frames: Activates the frame-based grid.

Snap offset to project start: Here the snap offset can be set relative to the beginning of the project. "Use current position for offset" specifies the current position as the grid's zero position.

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

Mixer setup

This item enables project and mixer settings to be configured, saved, and loaded.



Shortcut: **Ctrl + Shift + M**

Project and Mixer Configuration

Set the number of tracks, AUX buses, and submix buses. You can also define effects and signal routing for the tracks and the master section.

The window of the first track is always open for effects routing of individual tracks/buses. To adjust the settings for all tracks, click "All tracks" in the effects routing dialog.

Besides the routing button, two additional options are available:

- Use adv. dynamics in master for dyn./limiter: In the master section of the mixer, the resource-saving "normal" dynamics may be used as a dyn./limiter or for high-quality advanced dynamics.
- Automatic latency compensation: Activate this option for automatic latency compensation of DirectX and VST plug-ins. This is active by default.

Master Routing

- Keep current setting: The current setup in the project will not change.
- Stereo master: Normal mixing occurs via Samplitude's stereo master. If the sound card provides several devices, select one in the mixer (in the "out" bar below the volume faders) or in the play parameters (shortcut: "P").
- Surround master: If this setting is active, master channels are provided for 5.1 Surround mastering.

Detailed information about the Surround setup is available in "Surround sound -> Surround project setup (view page 225)".

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 Setup (Track Assignment, Routing)

Select the in and output device assignment for several tracks simultaneously in this area.

I/O Devices: Define the routing for playback, recording, or both.

Track number (0 for all): Start with track 0 as active track: This option permits a change to I/O routing for a certain number of tracks. If you set the track number 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 "Do not change the setup currently used in the project" 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.

Route 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 all tracks to different mono/stereo 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.

Initialize tracks on Surround channels: The corresponding tracks are created here after the Surround setup has been selected.

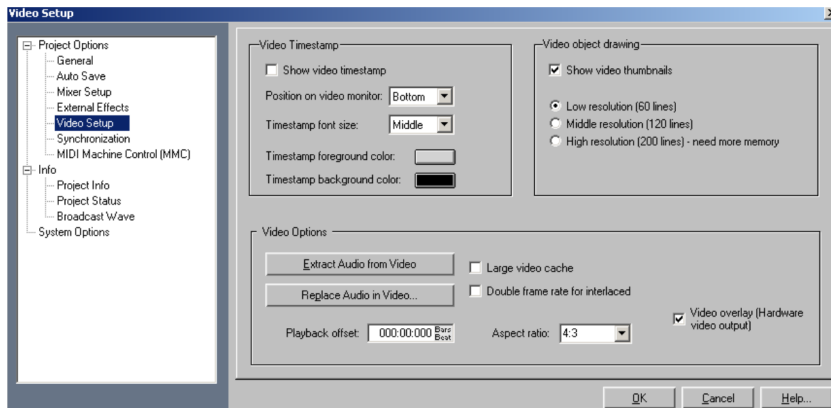
Use saved setup in preset: The settings saved in the preset are used.

External effects

Detailed information on this topic can be found in "Effects -> Breakdown and functionality -> External hardware effect integration".

Media link

The media link in "Options -> Project properties" allows you to link video and MIDI files with your Samplitude project to play them simultaneously.



More detailed information about the media link is available in "Synchronization -> Media link (view page 358)".

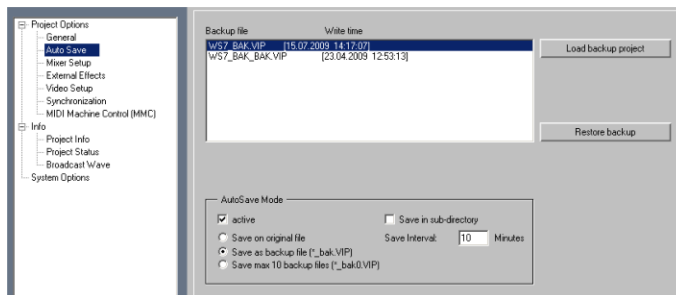
Playback Options

More information is available in "Playback -> Playback parameters (view page 578)".

Shortcut: P

Project Auto Save...

The **Autosave** dialog is available via the project options.



This dialog provides options for creating backups of projects. The button **View backup project** opens the selected backup in the arranger.

The button **Restore backup status** restores the project to the status of the backup. If the current project was changed previously and not saved, then the last status may be saved with the extension `_OLD.vip`.

Autosave Mode

Autosave mode may be activated with the corresponding check box. Select from the following options:

Save to original: Overwrites the original file in specific intervals with the current status. If the option **Save in subfolder** is selected, the previous status, i.e. the pre-previous status will also be updated in the **Backup** subfolder.

Add backup file (*bak.VIP): Adds a backup file and updates this at specific intervals. If the option **Save in subfolder is selected**, the backup file will be updated in the **Backup** subfolder each time it is saved.

Max. 10 backup files (*bak0.VIP until *bak9.VIP): Specify save intervals for up to 10 backup files and number them. If the option **Save in subfolder is selected**, the backup files will be updated in the **Backup** subfolder each time they are saved. After 10 backup processes, the final backup file will receive the extension *bak9.VIP, the oldest backup file will be deleted, and the others will move downwards in the sequence.

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...

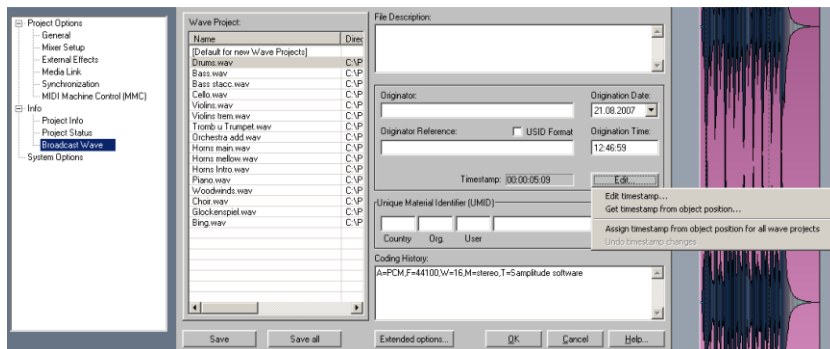
The broadcast wave manager extension makes it possible to store information about the audio file, i.e. metadata, in a chunk of the BWF file. This metadata may be proprietary, however we recommend observing the guidelines set out by EBU and SMPTE.

Wave Project

This displays a list of the data included in the project. Select the respective file using the mouse to display the individual information. The corresponding BWF information will be displayed in the right half of the window.

The main application of the BW manager, however, is defining the meta data included in the broadcast wave extension of the recorded audio material.

This metadata is then available to all users of the audio file in the future. Some applications can read or extract this meta data for further purposes, such as the administration of the audio data in the databases.



Broadcast Wave Manager - List of Individual Fields

All values that refer to the project are saved and applied to new audio data.

File Description: This is a freely usable text field. Enter a maximum of 256 ASCII 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 specified by the originator. This can be an internal reference number, for example. In the EBU range, a recommendation according to EBU R99-1999 specifies how this field should be constructed. To assign the entry of this property, activate the "USID" option. Next, format the entry according to EBU guidelines. Maximum of 32 characters.

Date: The date of the file's creation is displayed; this may be edited if audio material was saved for the first time as a file, even though the recording is somewhat older and the date needs to be valid as a reference. If the entry is edited in the BW manager, then a value independent of the file properties results.

Time: The time at which the file was created initially is shown here. Like the date, this is automatically created from the file properties, but may also be edited in the same way as mentioned above.

Timestamp: The timestamp saved in the BW (broadcast wave) 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 provides information about the time on the day of recording. The timestamp may be edited or applied from the object position. You can assign the timestamp from the object positions to all wave files.

Unique material Identifier (UMID): Handling of the UMID is regulated by SMPTE. An corresponding document is available from SMPTE. The code number is SMPTE 300M-2000. We recommend these reading the guidelines and agreements regarding the use of the UMID before using this feature, which may be applied according to your needs. Use of UMID is not absolutely necessary for a valid BWF.

Coding History: Besides 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 of this field contains a T value. This is a comma-free text string for entering serial numbers of recording tape machines, codecs, dither types, AD converters, or special signal editing applied to the file (like denoising).

An entry is made when a file is recorded in Samplitude. If this file is processed again (e.g. the bit width has been reduced or via MPEG encoding), then an additional entry is added.

Use of this coding history is regulated by EBU recommendation R98-1999.

Broadcast Wave Manager - Advanced Options

Save BWF graphic data in wave files: If this option is activated, Samplitude creates a so-called "peak chunk" that saves graphic information as a meta data in the file. Samplitude usually saves graphic data in separate files (.ho files).

Read BWF graphic data and use data in Samplitude: If this option is activated, the peak chunk included in a BW file is used instead of a .ho file.

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 792)".

Destructive Wave Edit Mode

This menu item activates or deactivates the destructive editing mode in the wave project window. If this point is deactivated, then Samplitude is put in "Virtual wave editing" mode.

Detailed information about destructive editing is available in the chapter "Working in the project window -> Samplitude as a wave editor (view page 108)".

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/DirectX/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 25)" for more information.

Buffer Settings

Please also read "System settings -> Audio setup (view page 25)" for more information.

Device Address/Driver Communication

Please also read "System settings -> Audio setup (view page 25)" for more information.

Monitoring Settings

There are different ways of monitoring. These settings are used as presets for new projects.

For more information about monitoring, please see "System settings -> Monitoring settings (view page 27)" for more details.

Audio Device

More information about this is available in "System options -> Audio devices (view page 31)".

MIDI

For more details about MIDI settings, see "System settings -> MIDI settings (view page 31)".

Hardware Controller

Detailed information about this is available in "Hardware controller".

Metronome options

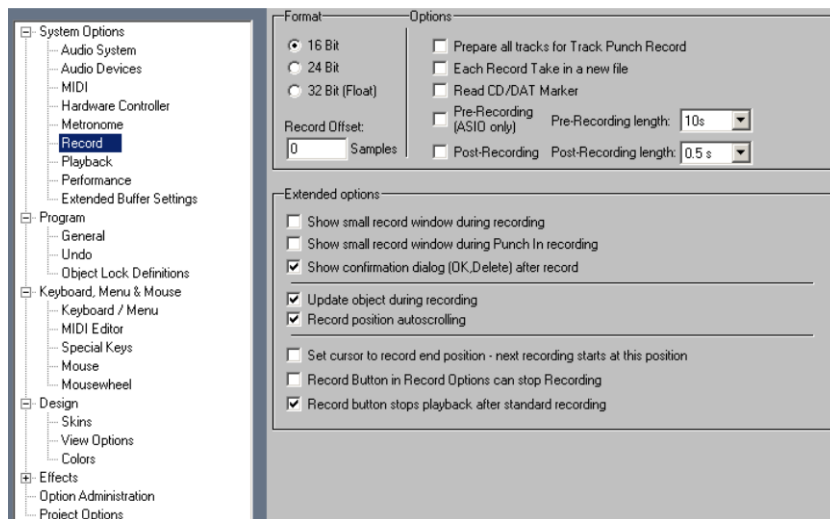
Detailed information about this is available in "File -> Program Preferences -> Metronome Options (view page 84)".

Record

This dialog enables the selection of various dialog configurations and cursor behavior 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 then available:

- **Pre-activate all tracks for track punch recording**

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.

- **Save each take in a new file**

This option saves each take of a recording as a new file. Loop recordings, however, are collected as a take and then saved into the same 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). This recording option reads this marker information from the SPDIF input of the sound card and applies it to the VIP, provided that the selected audio device supports this.

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. Prerecording lengths of 2, 5, 10, 30, 60, and 120 seconds may be set.

If a recorded object is dragged to the left, then the audio material recorded before actual recording start is visible according to the set pre-recording time.

Post recording: Post-recording may 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 the end of the object to the right after recording is dragged out, the portion of extra material recorded will be displayed. The preset for post-recording equals 0.5 seconds.

Advanced options

Show small record window during direct recording: Displays a small non-modal recording window with the most important recording controls while recording.

Show small record window during recording/punch in recording: Displays a small non-modal recording window with the most important recording controls during punch recording.

Show confirmation window after recording: If this option is checked, a dialog will appear after the recording process to choose whether to keep or discard the recording.

Note: Select the option "Display confirmation window after pressing the record button..." , and the recording will always be saved directly after it has completed. On the one hand, this offers the best protection 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 until they are manually deleted.

Update object during recording: This option increases the recorded object constantly along with continuous recording.

Autoscroll recording position: If this option is activated, the window will scroll along with the recording cursor.

Set cursor to record end - start next recording at this position: If this option is active, the play cursor is moved on recording; playback/recording then starts from this position.

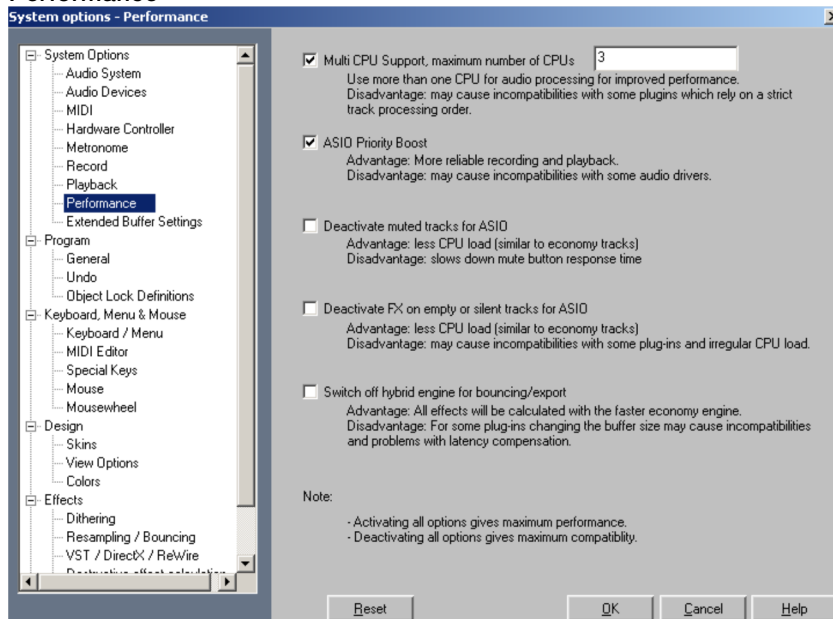
Recording button in recording options can stop recording: Specify that the "Record" button in the **Recording options** (shortcut: **Shift + R**) is able to start or stop the recording process.

Record button ends playback after standard recording: If this option is activated, then the running recording and playback may be ended by pressing the "Record" button in the transport console.

Playback

More information about playback settings is available in "Playback -> Playback parameters (view page 578)".

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.

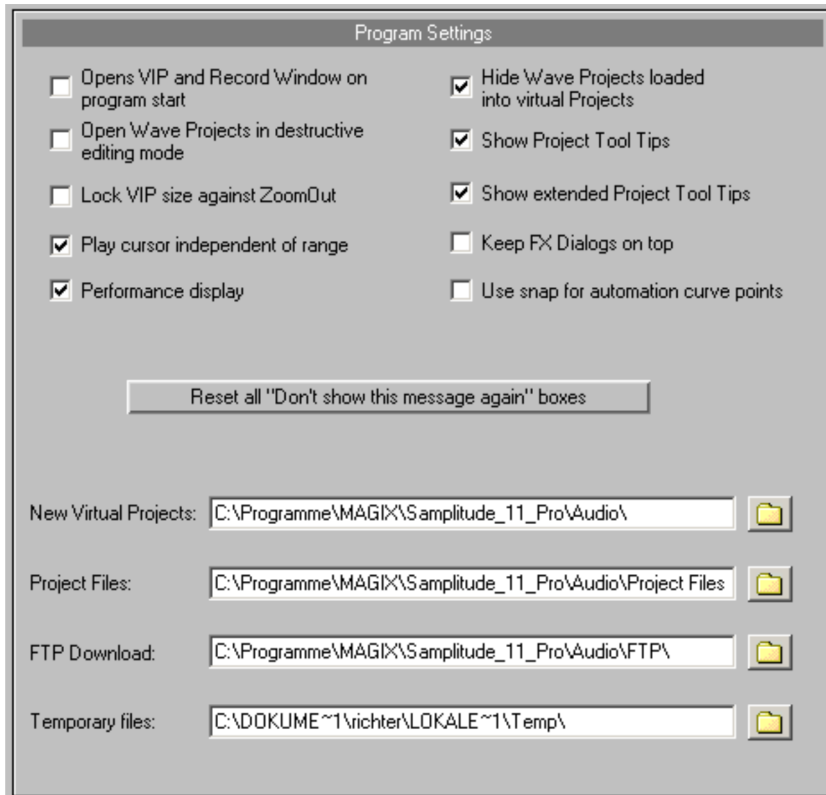
Switch off Hybrid Engine for bouncing/export: This option calculates all effects during the bouncing/exporting process using the faster "Economy Engine". This may cause irregularities when some plug-ins attempt to compensate for latency.

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 500)".

Program General



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 active, wave projects will be opened in destructive 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.

Play cursor independent of range: This option allows the play cursor 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. If the display shows values of 100% or more, then the processor is overloaded. In that case, take the following steps:

- Reduce the number of real-time 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) up to 16,000 or 32,000 samples.

Note the message displayed in the status bar in case overload occurs during real-time previewing of complex effects like the Denoiser or FFT analyze filter.

Hide wave projects loaded into virtual projects: Hides windows in wave projects that are loaded directly into the VIP.

Show project tool tips: If this option is active, additional information is displayed when the mouse pointer hovers over an interactive element.

Show extended project tool tips: If this option is activated, the extended tool tips are displayed if the mouse pointer is held over an inactive element.

Keep FX dialogs on top: This option keeps all effects dialog windows "always 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: Press 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 under this path.

Project files: All other wave files which are to be saved on the hard drive and that cannot be allocated to a specific project are saved under the specified path.

FTP download: All files downloaded via the integrated FTP client will be saved under the specified path.

Temporary files: This preset path is directed to the standard temporary files folder. Make sure 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 497)".

Locking objects

Detailed information about this is available in "File -> Program preferences -> Lock Definitions".

Keyboard, Menu & Mouse Multi Track mixer



Dual Row Recorder



Keyboard

console.
interface.

Recording Mixer



track
view in

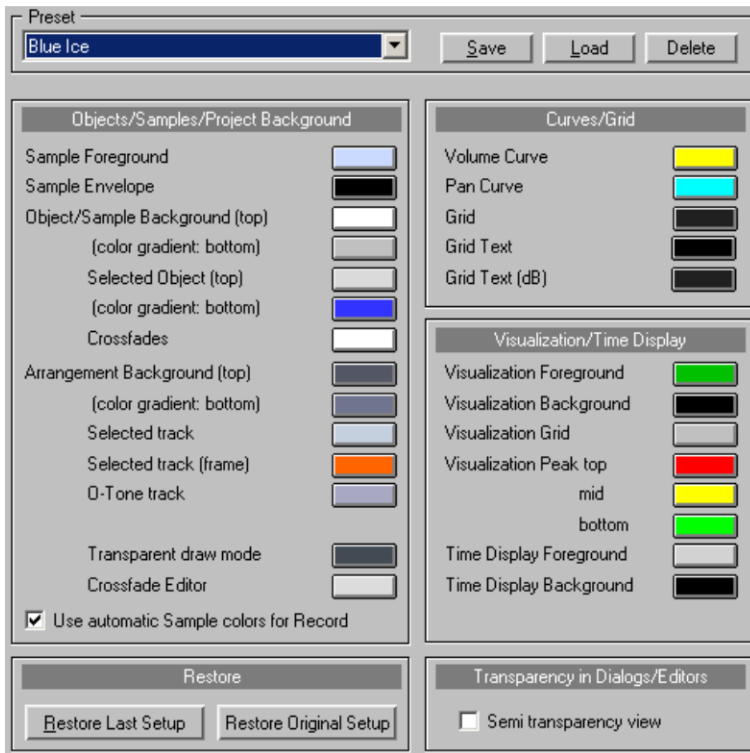
Various skins may be selected by clicking on the icon to the top left in the title bar of the Mixer (view page 158) or in the Menu bar of the Arranger (view page 47).

System options > Design > View options

Detailed information on display options can be found via "File > Program Preferences > Definition... (view page 491)".

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 498).

Resampling/Bouncing

Specify the settings for record resampling (normal, high, very high), playback resampling (normal, high), and object resampling for new objects (normal, high) as well as the resampling quality during bouncing (high, very high, ultra high 1, ultra high 2).

You can also select the following freeze options here:

- Keep mono, if possible
- Use additional samples for object freeze
- Object freeze without object volume

16-bit, 24-bit, 32-bit formats are available

VST/DirectX/ReWire

Detailed information is available in the section "Mixer -> Effects routing/plugin-ins dialog -> VST/DirectX/ReWire setup plug-in buffer (view page 176)".

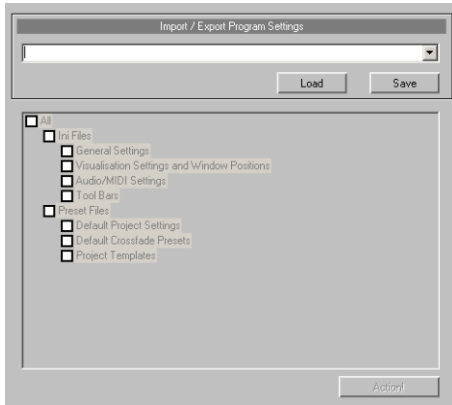
Destructive Effect Calculation...

Choose in the advanced destructive effects calculation options (view page 601) 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 administration

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.



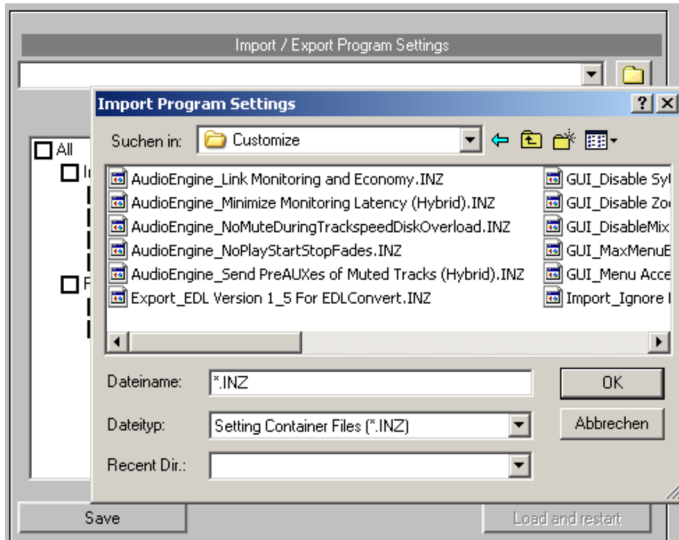
The program settings are saved as "**Settings Container Files (*.INZ)**".

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 479)
- Settings displays and window positions (view page 491)
- Audio/MIDI Options
- Toolbars
- User administration (affects all users and user settings) (Sequoia only)
- Standard project settings (view page 33)
- Standard Crossfade-templates (Sequoia only)
- Project templates (view page 449)

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.

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.

Project options**General:**

Please also read "System settings -> Project Options -> General (view page 33)" for more information.

Auto save:

Detailed information about this is provided in "File -> Project Properties -> Project Auto Save (view page 470)...".

Mixer setup:

Detailed information is provided in "File -> Project Properties -> Mixer Setup (view page 468)...".

External effects:

Detailed information on this topic can be found in "Effects -> Breakdown and functionality -> External hardware effect integration".

Video setup:

Detailed information is provided in "File -> Project Properties -> Media Link...".

Synchronization:

Detailed information is provided in "Synchronization (view page 345)".

MIDI Machine Control (MMC):

Detailed information is provided in "Synchronization (view page 345)".

Project information:

Detailed information is provided in "File -> Project Properties -> Project Information (view page 471)...".

Project status:

Further information is provided in "File -> Project Properties -> Project Status (view page 471)...".

Broadcast wave:

Detailed information is provided in "File -> Project Properties -> Broadcast Wave Manager (view page 471)...".

Varispeed and Scrub Settings

This dialog enables tempo changes for virtual project playback to be made. Right click the scrub control button in the transport controls.

Sample rate: The preset project sample rate is displayed here. To change this value, a request window will appear to adapt all audio objects to the new sample rate as needed. This adjustment may also be made by moving or resampling the audio material. MIDI objects may also be adapted to the new sample rate. These may also be moved, but the musical position will remain the same.

Autoscroll/scrubbing: See "Playback parameters (view page 578)".

Stop at position: This stops the play cursor at the current position when the "Stop" button is pressed and doesn't jump back to the original position.

Device: Specifies the driver of the sound card for playback.

Active: Checking this box activates Varipitch/Varispeed.

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 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: This specifies the sample rate for Varispeed calculation. At a sample rate of 44,100 kHz, a value of 22,050 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.

MIDI Options...

More details about MIDI settings are provided in "System settings -> MIDI settings (view page 31)".

Synchronization Active

This command activates synchronization.

Shortcut: G

Synchronization Setup...

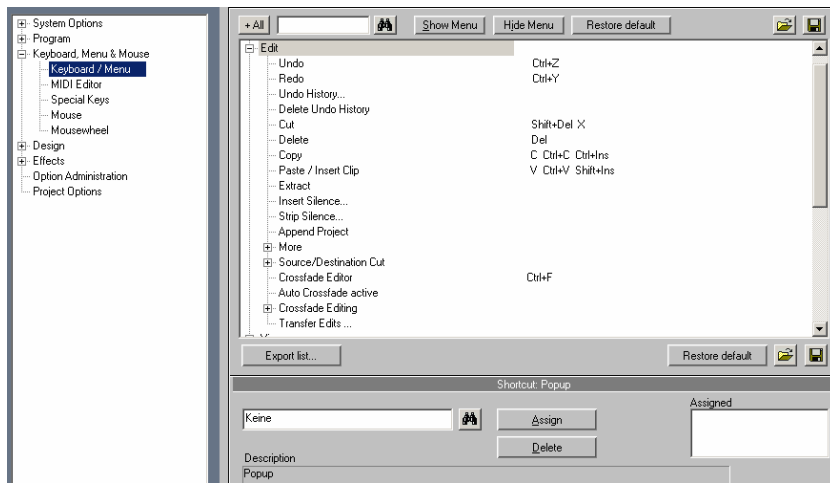
Shortcut: Shift + G

Read "Synchronization (view page 345)" for more information about synchronization settings.

MMC Setup...

Read "Synchronization (view page 345)" for more information about synchronization settings.

Shortcuts and Menu Settings



This dialog allows you to specify keyboard shortcuts for every menu function in Samplitude. It's possible to set the commands you use most often to a special key combination in order to open them quickly.

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

When you close Samplitude, the shortcut settings are saved in the Windows folder file "Sam_d.ini" so that they are available automatically the next time the program is opened.

The display of the entire menu tree of Samplitude is the key function of the dialog.

Here, you can select which menu point you want to create a new shortcut or if the menu point should appear in the main menu of Samplitude.

Displaying and searching for menu items

The menu is displayed in tree form, and submenus may be opened by clicking on the "+" symbol. You can also search for a specific menu point by entering any desired search term into the upper input field and then clicking on the magnifier next to it.

Show/hide menu item: Select a menu item which you would like to hide. **"Hide menu point"** removes the menu item from the menu. It will be shown in grey in the dialog's tree diagram. Please note that the menu point will no longer be selectable via previously defined keyboard commands. **"Show menu item"** activates the hidden menu item again. **"Reset"** returns everything to the original condition with all commands visible again.

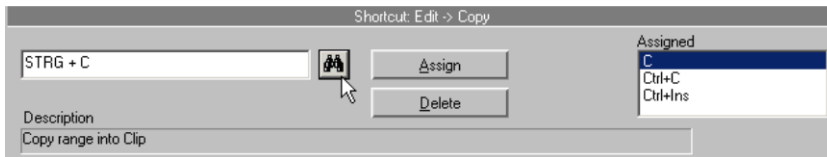
Load/save/reset: These buttons load, save, and reset the menu definitions, respectively.

Create keyboard shortcuts: Click on the desired menu point and then on the entry field underneath the "Shortcut..." bar. Then, press the desired buttons for the keyboard combination. Drücken Sie nun die gewünschten Tasten für das Tastaturkürzel.

Combinations of any key with "Shift", "Alt", or "Ctrl" may also be used. You can now assign the new shortcut with the button at the far right. Please do not use the space bar, "ESC", or "Ins" key (0 in the number block), since these keys are permanently assigned in Amplitude and cannot be changed.

Assign: This button activates the entered key combination as a new keyboard shortcut.

Find keyboard shortcuts: Click the input box underneath the "Shortcut..." bar. Next, press the keyboard combination buttons for the search. Click on telescope symbol on the side to view the menu command assigned to the keyboard shortcut.



Delete: Deletes the selected shortcut.

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

MIDI editor/Special buttons/Mouse wheel/Mouse

"MIDI editor" finds or assigns new keyboard shortcuts for editing MIDI events.

"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 78)": 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 78)": 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 137) with the mouse as long as the assigned shortcut combination/key combination is held down.
- Temporary switch for Snap mode (view page 66): Activates the "Snap" mode as long as the assigned keyboard shortcut/key combination is held down.
- Temporary switch for Object (view page 69) mode: Activates "Object" mode from "Range" mode as long as the assigned key is held down.
- Temporary switch for Curve (view page 70) 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 (view page 579)" mode: Activates "Preview" mode as long as the assigned keyboard shortcut/key combination is held down.
- Temporary switch 2 for "Scrub (view page 579)" mode: Activates "Preview" mode as long as the assigned keyboard shortcut/key combination is held down.

- Temporary switch for "Zoom mode (view page 101): Activates zoom mode as long as the assigned keyboard shortcut/key combination is held down.
- Switch for "Scrub mode (view page 107): Switches to "Scrub" mode indefinitely until the next time play stops.
- Temporary switch for "Cut mode (view page 70): 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" 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 for double mouse clicks**
- **Zoom lasso allows vertical zoom without "Shift"**
- **Disable mouse zoom on grid**
- **Old shift click behavior in mixer:** 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.
- **Treat knobs like 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 threshold:** When an object is selected by clicking it, it could be moved unintentionally. If "Move threshold" is activated, Samplitude will wait a set time until performing the operation.

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

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" option. 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.

Edit Toolbars

Samplitude lets you customize toolbars however you like. The corresponding editing dialog may be opened by double clicking any toolbar.

- Select an available button and press "**Insert**" to add the selected button to the toolbar.
- Select an available button and press "**Del**" to remove the selected button from the current toolbar.

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.

Font Selection...

Select a font for text display in Samplitude.

Font for Time Display...

Select a font for the time display here. The time display is opened via the "Window" menu.

Font for Manager...

Choose a font for the managers.

Metronome Options...

Detailed information about metronome settings is provided in the chapter "Samplitude quick start -> Workshop: recording -> Metronome settings (view page 84)".

Draw Setup

Definition...

This dialog can be used to change between the two viewing options "**Drawing mode 1**" and "**Drawing mode 2**". Wave form display, object, VIP, and wave form color categories can be edited.

Shortcut: Shift + TAB

Mode 1/Mode 2/Switch Modes

Select a view mode here, or switchback and forth between the two viewing modes.

Shortcut: Tab

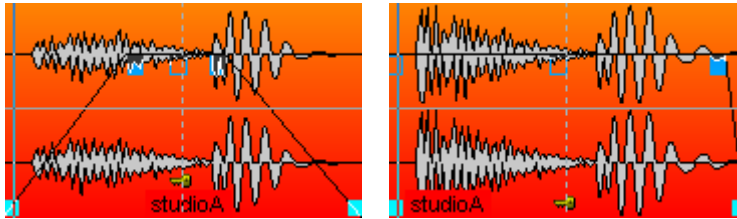
Waveform Display

Draw wave form: Turn the wave form display on and off here. When deactivated, you can see volume and panorama curves more clearly.

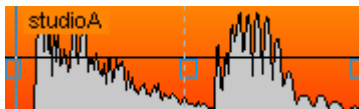
Note: In "Drawing mode 2", the standard default setting for the wave form display is switched off. This may be changed at any time.

Scale with fades/curves: In this default display mode, the wave form overview is scaled according to the settings for the fade in/out or volume curves. This makes the decay of the

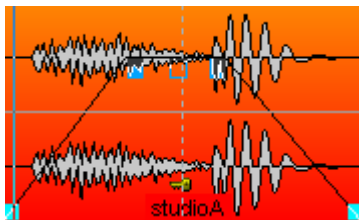
sound during a fade out or for volume curve changes more visible. This mode also offers good visibility when crossfading audio material.



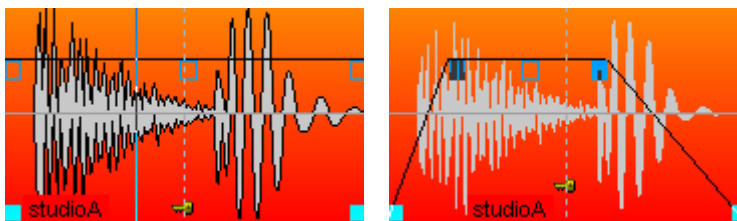
Halved waveform: Switches the graphical representation of the samples to half the waveform display size.



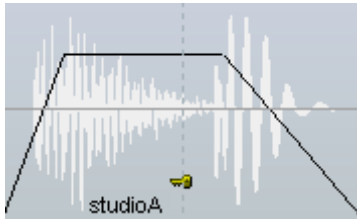
Separate stereo: If this option is active, stereo tracks are displayed separately one above the other.



Draw envelope: If this option is active, an outline is drawn around the envelope of the waveform.

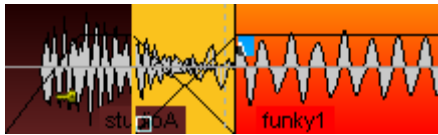


Gray out muted objects/tracks: Muted objects and tracks will be shown gray.

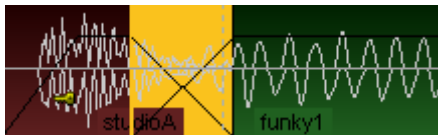


The second column is especially useful for a more clear crossfade display.

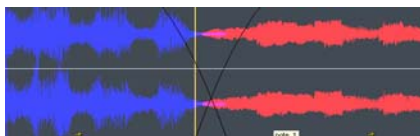
Standard: The waveform of the second object is drawn over the waveform of the first object:



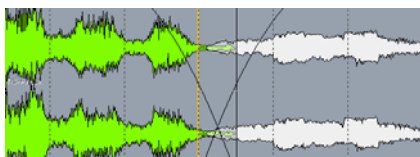
Envelope only: When this radio button is selected, only the waveform outline (envelope) is drawn. Both objects are well visible in crossfades:



Transparent: If this button is selected, the colors of both crossfade waveforms are overlapped. The waveforms appear against a dark background. We recommend combining this and the "Waveform colors -> Red/blue alternation" to optimize the display of crossfade areas:



Interleaved: When this option is selected, a sample of the left object and a sample of the right object will be drawn. You may then visually assess the fade area, a task made especially easy if the adjacent objects have different colors:



VIP

Various VIP components (buttons/sliders, peak meters, show border, arrangement text, compressed waveform, and CPU peak meter) can be activated/deactivated here.

Buttons/slider displays the track header with all mixer controls, i.e. the channel faders, solo buttons, etc. If you always have the track editor open, then you won't have to use these functionally identical control elements at this location. However, if the track header is left open, then only the track header's **peak meter** will be able to be hidden.

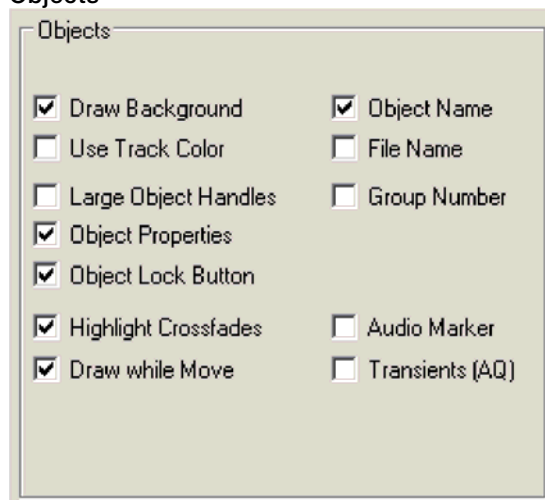
If you **hide the borders**, the edge of the selected track will disappear.

Arrangement text identifies the tracks for the two editing ranges in "Universal" mouse mode. If you place a check mark here, you will see "Play cursor and range manipulation area" in the top half of the track and "Object manipulation area" appears beneath this.

Compressed wave display: This display option causes the waveform to be distorted according to the zoom level. The transition from silence (less than -90 dB) to signals with lower levels (e.g. noise) is also shown more clearly.

CPU peak meter: If this option is activated, the **peak meter in the track header** displays the CPU load per track. The upper peak meter displays the load on the Economy Engine for the left channel, while the peak meter below this displays the load on the Low Latency Engine for the right channel. In this case, a display value of 0dB equals a load of 100%.

Note: The dB displayed by the track header peak meter depends on the skin (view page 481) selected for Samplitude.

Objects

Draw background: This activates the background color view of the selected objects. Every object may be assigned its own color (see "Object .> Object color/name (view page 563)" or in the object editor under "Color").

Use track color: The track color will be applied used as the background color.

Large Object Handle: 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 block 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 549).

Highlight crossfades: With this setting, the crossfades between the objects are displayed in contrasting colors.

Draw while moving: The user interface is updated while moving the mouse. This option can be deactivated for computers with low processing capabilities.

Handle for fade overlap: Use this handle to set the offset ratio for crossfades without having to open the Crossfade Editor or Object Editor.

Additional information about object representation:



Object Name: Shows the object name.

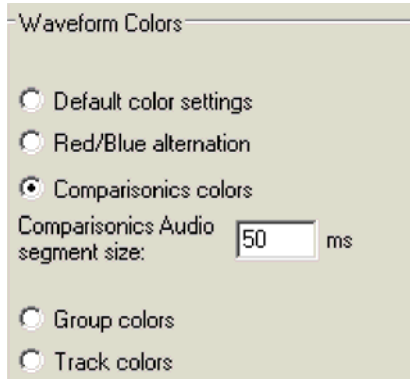
File name: The file names will be displayed.

Group number: Objects can be grouped in VIPs. The groups are numbered. When displaying these group numbers a quick overview appears showing which objects belong to which groups.

Original position: This shows the original recording position at the beginning of the object in the set time code format. This way, a comparison of values in the object with the position on the timeline or on the time code of the picture helps clear up possible deviations in the synchronization.

Audio Markers: Use this option to display markers in objects that are present in the wave projects that are based on them.

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 > System/Audio -> Design -> Colors (view page 482)".

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 may be optimally viewed in the "Transparent" and "Stacked" drawing modes.

Comparisionics colors: This display mode uses a patented algorithm from Comparisionics™ for coloring the display of the sound material. Depending on the pitch and other parameters, a special color tone for each segment of audio material can be calculated and displayed. Deep sounds get blue color tones; high sounds get green/yellow/red colors. Tonal material appears clear, noisy material features gray tones. This makes it easy to recognize different pitches optically, for example a saxophone solo versus unwanted noise.

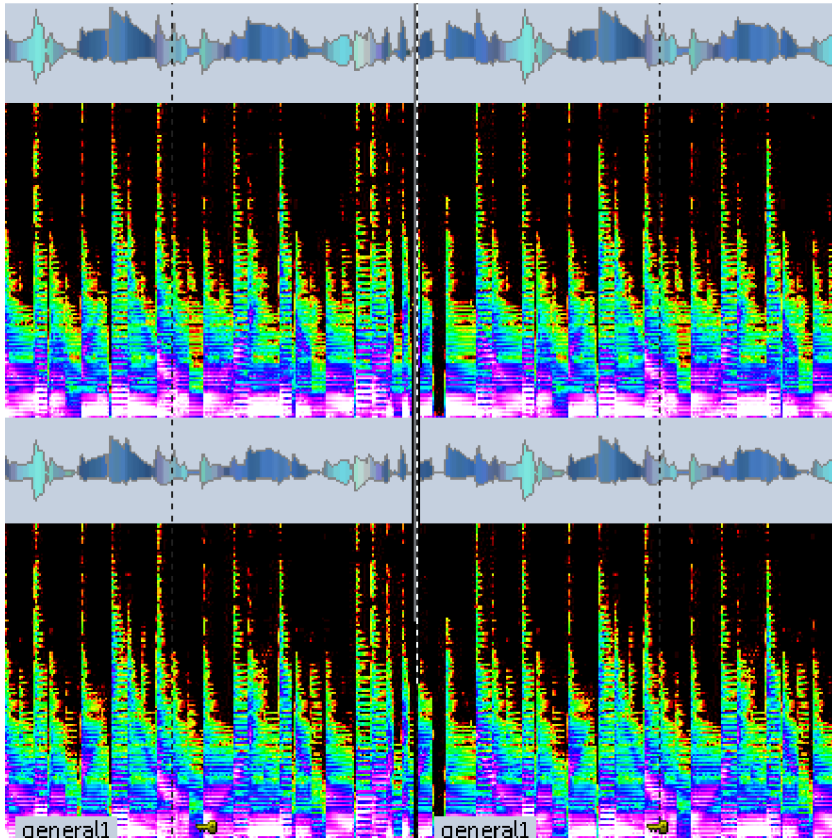
The segment size for color calculation can be entered in "**Comparisionics audio segment size**". Sizes 50 ms or longer are recommended so that there is enough material for color determination available within the segment.

On the basis of the Comparisionics colors, a completely new audio search function is available via "Range -> Comparisionics audio search (view page 571)"; with these help ranges, identical or similar audio material in the project may be found quickly and easily. More information on Comparisionics algorithms can be found at www.comparisionics.com.

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 colors palettes. The graphical data that has been created will be saved in an extra *.hs file.



The graphical interpretation of the music is displayed in the Spectral display using 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 482)".

Undo Definitions...

Use this editor to switch the undo function on and off for virtual projects, wave projects, and plug-in settings. The amount of undo steps may also be entered. A value of 20 means that the last 20 changes to a project may be undone.

If you check the corresponding box, temporary undo files be created as HD wave projects for RAM wave projects.

Dithering Options...

How does dithering work?

An audio signal is quantized at a higher or lower resolution with every A/D conversion. During quantization, the signal receives an echelon form via limiting of the possible amplitude values. An 8-bit signal, for example, possesses only 256 amplitude values. This echelon formation causes distortion and thereby deforms the signal, especially impairing the sound of the lower levels.

Dithering can be understood as "mixing in" of low-level noise which seriously reduces the human ear's detection of sound defects.

When to dither?

Samplitude always dithers a signal in integer format when it is saved or exported.

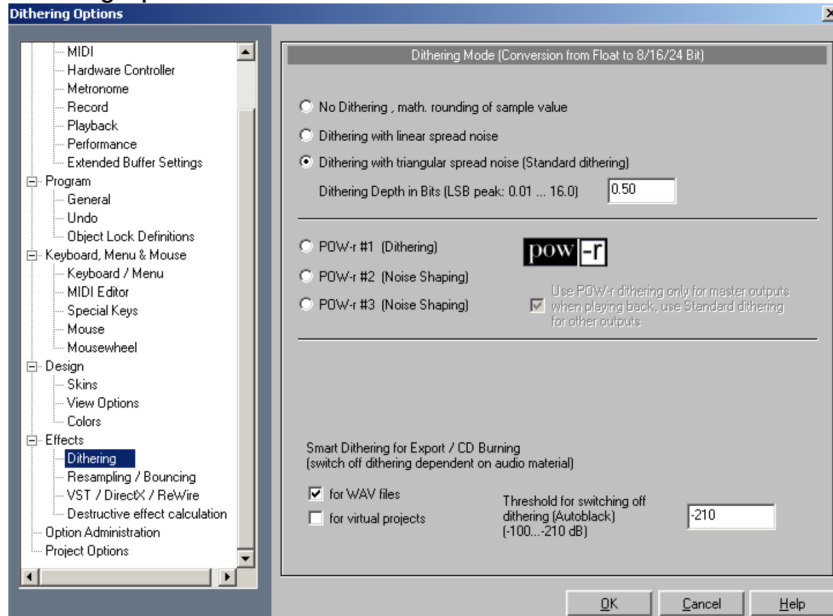
This is the case for the following situations:

- 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. Dithering will only occur on 16-bit exports and when burning CDs if the bit-depth is 16-bit. If the original material is 16-bit, dithering only occurs when there are modifications to the bit depth. 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.

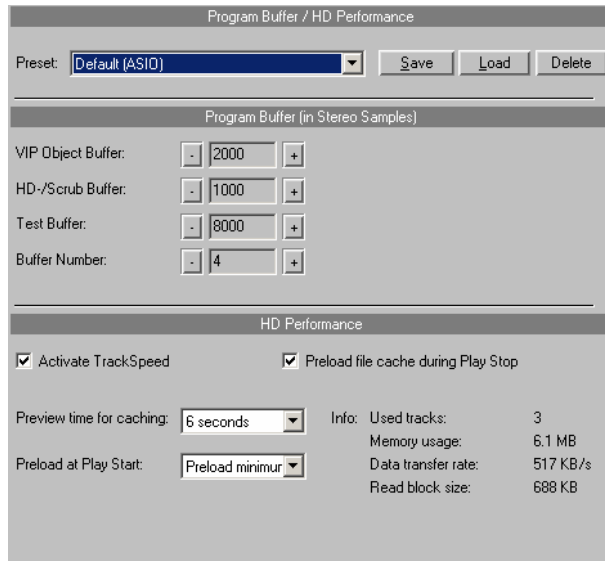
Threshold for switching off dithering (autoblack): Specify a threshold value below which any dithering noise that is created should be muted. The range of values in this case is between -100 dB and -210 dB.

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 no drop-outs or scratches are heard during playback, you don't have to make any changes.

Program buffer/ hard disk performance

For special cases, you can set and save your own buffer settings. In addition, special presets are available, e.g. for scrubbing and ASIO applications.



Program Buffer (in stereo samples)

VIP Object Buffer: Since error-free playback is usually more important than fast reaction times, this value should be increased when playing multiple tracks. This is the only setting relevant for playback and editing in virtual projects.

HD/Scrub Buffer: This buffer is put useful during direct playback of HD wave projects. You can try out smaller values here to achieve faster reaction times.

Test Buffer: This buffer is only used for previewing the effects from the effects menu in real time.

Buffer Number: Specify how many of the buffers described above should be used here. More buffers increase reliability, but also increase the memory requirements. Reaction times become longer. You can check the current buffer usage levels in the status bar to 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 drives 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 reducing performance. This makes short reaction times between user activities possible, for example with the mixer and various object handles.

There are several settings, and TrackSpeed can be customized to your individual needs.

Activate TrackSpeed: Set a check mark here to activate TrackSpeed.

Load File Cache in Stopped State

If this option is activated, the file cache will be preloaded each time the play cursor 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 RAM.

Read Ahead: 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). Large values delay start, but increase play stability when track count is high.

The dialog also displays the following project information:

- Tracks used
- Total memory requirement
- Data transfer rate
- Reading block size

Resampling Quality Options...

Use these options to set the following resampling/freeze calculations.

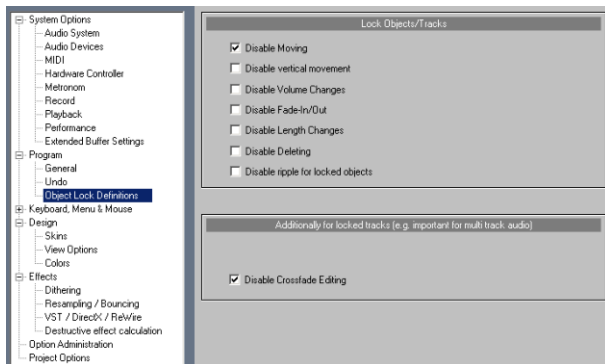
Resampling options

- 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)

Freeze options

- Keep mono, if possible
- Use additional samples for object freeze
- Object freeze without object volume
- Format (16-bit, 24-bit, 32-bit)

Locking options



Select which functions should be disabled by locking objects or tracks here. You can select the following options:

Locking objects

Disable moving: Prevents objects from being moved horizontally. This avoids unintentional offsets between individual tracks for multi-track recordings.

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.

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 601) 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.

More

Disconnect Internet

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.

Exit

The program Samplitude is being shut down. All projects changes will be lost if "Save" is not selected.

Recent File

In the last section of this menu the 10 projects opened last will be displayed.

Edit Menu

This menu features functions that are applicable only to wave projects ("Edit -> More -> Copy to"), and others are only for virtual projects ("Cut silence").

Edit functions like "Extract" require range to be selected; 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 bar (view page 69).

In addition to the various mouse modes, numerical values in the dialog windows can be changed by clicking on the input field or by moving up and down while holding down the left mouse button. This turns the play cursor into a vertical double-ended arrow.

"Universal" mode

This is the preset mouse mode for Samplitude. All necessary functions can be reached with the left mouse button. Right-clicking always opens a context menu.

Detailed information about "Universal" mode is available in "Screen elements -> Mouse mode bar -> Universal mode (view page 69)".

Left mouse button:

Upper half of track

Drag: Open and move ranges.

Click: Place start position of play cursor.

Lower half of track

Click: Selects objects.

Click + Shift: Selects multiple objects (including all objects between two clicked objects).

Click + Ctrl: Select multiple objects.

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

Drag + Ctrl: Duplicate one or more objects.

Drag + Shift + Ctrl: Duplicates one or more objects on a track; the position is maintained.

Double click on object: Double click opens the object editor

Lasso function: Clicking next to an object while dragging to the right activates the object lasso to select multiple objects.

Clicking next to an object, while left-dragging activates the object lasso for selecting multiple volume or panorama events.

Object handles

Middle handle (top): Changes the object volume. The exact value in dB can be seen 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 handle: Sets the start and end position of the object.

Volume and panorama curves:

Double-clicking on a curve: Creates a new curve event, while double-clicking again deletes an existing event.

Click on a point: Selects a point.

Click on an event + Ctrl: Allows selection of multiple events.

Click on an event + Shift: Multiple selection of start and end events, including all curve events in between.

Drag a selected point: Moves the selected point.

Drag an event + Ctrl: Moves multiple selected points.

To delete multiple events, select them and choose "Edit -> Delete curve handle" or simply press the "Del" key on your keyboard.

Right mouse button: Context menu

Range mode

In this mode, only ranges and the play cursor can be manipulated.

Detailed information about "Range" mode is available in "Screen elements -> Mouse mode bar -> Range mode (view page 69)".

Left mouse button:

Drag: Stretches and moves ranges with the "Shift" key.

Clicking beside range: Sets the start position of the play cursor.

Held down "." period key: Switches temporarily to "Object" mode to move objects.

Held "-" dash key: Switches temporarily to "Curve" mode to move and edit volume and panorama curves.

Right mouse button: Context menu

Curve mode

This mode allows Panorama and volume Curves to be drawn, edited and deleted with the left mouse button.

Read more on this in the "Screen elements -> Symbol bars > Mouse mode bars (view page 70)" section of the manual.

Left mouse button:

Drag to left or right: Dragging the lasso selects multiple volume or panorama automation curve events.

Click once on a curve event: Creates a new event; a double click deletes it again.

Click on a point: Selects a point.

Click on an event + Ctrl: Allows selection of multiple events.

Click on an event + Shift: Multiple selection of start and end events, including all curve events in between.

Drag a selected point: Moves the point.

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

To delete multiple events, select them and choose "Automation -> Delete curve points", or simply press "Del" on your keyboard.

Right mouse button: Context menu

Object mode

In this mode, objects may be moved and curves may be edited with the left mouse button.

Detailed information about curve modes is provided in "Screen elements -> Mouse mode bar -> Object mode (view page 70)"

Left mouse button:

Click: Selects objects.

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

Click + Ctrl: Select multiple objects.

Drag 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: Duplicate one or more objects.

Drag + Shift + Ctrl: Duplicates one or more objects on a track; the position is maintained.

Double click on object: Double click opens the object editor

Lasso function: Clicking next to an object while dragging to the right or left across multiple objects activates the lasso.

Object handles

Middle handle (top): Changes the object volume. The exact value in dB can be seen in the tooltips.

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

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

Right mouse button: Context menu

"Object and curve" mode

In this mode, you can move objects and edit curves using the left mouse button.

Left mouse button:

Click: Selects objects.

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

Click + Ctrl: Select multiple objects.

Drag 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: Duplicate one or more objects.

Drag + Shift + Ctrl: Duplicates one or more objects on a track; the position is maintained.

Double click on an object: Opens the object editor.

Lasso function: Clicking next to an object while dragging the lasso to the right across multiple objects selects the objects within the loop. Clicking next to an object while dragging the lasso to the left selects multiple volume or panorama curve events.

Object handles

Middle handle (top object edge): Changes the object volume. The exact value in dB can be seen in the tooltips.

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

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

Click on a curve: Creates a curve event; a double click deletes it again.

Click on a point: Selects a point.

Click on an event + Ctrl: Allows selection of multiple events.

Drag a selected point: Moves the selected point.

Drag an event + Ctrl: Moves multiple selected points.

To delete multiple events, select them and choose "Automation -> Delete curve points" or simply press the "Del" key on your keyboard.

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, the left one controls manipulations of the range.

Left mouse button:

Double click on volume and panorama automation curves: Creates a new curve event; another double click on the same curve event deletes the event.

Right mouse button:

Click: Selects objects.

Click + Shift: Select multiple objects.

Drag objects: Moves objects

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

Drag + Ctrl: Duplicate one or more selected objects.

Drag + Shift + Ctrl: Duplicates one or more selected objects on a track; the horizontal position is maintained.

Click on a point: Selects a point.

Click on an point + Shift: Allows selection of multiple points.

Drag selected points: Moves several events.

Wave (HDP/RAP) Mouse Mode

Range

Use the left mouse button to set the play cursor 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

Left click: Click on an object to separate it at the corresponding position.

Right mouse button: Context menu

Pitchshift/Timestretch mode

The object handles below and in the centre can be used for directly setting the playback speed and pitch.

The timestretch Mouse Mode simplifies working with tempo markers.

Create **BPM markers** (Tempo changes) and **by directly clicking the desired playback position and holding down Shift**. You can adapt the tempo by moving the mouse vertically while holding down "Shift".

You can create **Grid position markers** in "Timestretch" mouse mode by clicking the desired playback position and holding down "Alt". Also you can shift the snap by moving the mouse horizontally.

You can create **Grid position markers** with "**Alt**" + **horizontal mouse movement without adjusting time positions**, e. g. to manipulate the bar grid and to adjust it to available audio or MIDI objects.

Left mouse button:

This mode is only different from "Universal" mode because of the handle function (the five boxes on the selected objects):

Middle handles: Object pitch may be changed within a range of +/- 6 semitones. The process used may be selected in the object editor.

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

Lower right handle: This handle controls the length changes made to an object via timestretching. The same time section (starting length of an object in the wave project) may be extended or shortened via timestretching. Select the stretching mode used in the object editor.

Right mouse button: Context menu

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 of the embedded plug-in beforehand with the shortcut "**Ctrl + Alt + move the corresponding plug-in's handling 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 in the track you can draw in the curve of the selected parameter using the pen in Automation Draw mode.

The selected parameter's curve can be seen in the track in the foreground; all other curves are optionally displayed as thin lines via the context menu.

Tip: You can also move any plug-in parameter at any time via the small horizontal faders in the track editor's automation box while using "Automation drawing" mode (e.g. touch, latch, overwrite, or trim). Latch, Overwrite oder Trim) aktiviert haben.

Left mouse button: Freehand draw function for automation curves.

Right mouse button: Context menu

Spectral mode

With the "Spectral mode (view page 72)" you can remove noise from a track object, without the wanted signal being audibly influenced. Editing can be done directly in the Arranger window.

After calling up "Spectral mode" highlight the unwanted noise by drawing a rectangle around it using the mouse. With help from the handle you can adjust the selected area. 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

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.

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

Right mouse button: Context menu

Zoom mode

Left mouse button: zoom in

Right mouse button: zoom out

Color mode

Left mouse button: Changes the background color of the object

Right mouse button: Changes the object's wave form color

Object Mode

Detailed information about object modes is available in "Screen elements -> Toolbars - overview -> Object modes (view page 77)".

Undo

Amplitude 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 497)".

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.

Insert effect	12:33:49
Button/Fader	12:33:36
Load VIP	12:32:59

In this dialog, you can return to an earlier work stage. This list can be removed in the "Edit -> Delete undo levels (view page 513)".

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 (view page 427), like the virtual projects, does not contain the actual audio files, just the links to them. The VirtClip contains the same count of tracks as the selected area.

Shortcut: X

Wave projects

The data of the selected range is copied into the clip (view page 437) 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 play cursor 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 Marker Positions

The object is split into pieces at the marker positions that lie within the object borders. The newly created objects will be named after the previous markers.

Split Objects at Track Marker Positions

The object is split into pieces at the marker positions that lie within the object borders. The newly created objects will be named after the previous track markers.

Split Objects at Audio Marker Positions

The object is split into pieces at the audio marker positions that lie 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 (view page 426) or VirtClip is inserted at the current position of the play cursor (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 play cursor 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)

With this command you can delete selected objects from the virtual project. For selected ranges, objects as well as track automation curve points above them will be deleted.

In connection to the button "Link objects on 1 track" in the button bar, the objects lying behind the deleted area of the corresponding track are pulled up, shortening the total length of the track. In other object modes, the range is replaced by silence.

Use "Object -> Cut objects -> Delete objects (view page 546)" to remove the objects from the range.

You can delete any individual automation curve handle by selecting it in the project window using the mouse and going to "Edit -> Delete curve points".

Shortcut: Del

Wave projects

The files of the selected area will be deleted and the segment of the wave project located after the area will be pulled into its place.

Caution: If you are using offline Audio edit mode and have deactivated the "Undo" function ("Options -> Program preferences -> Undo Definitions (view page 497)"), the files cannot be restored.

Delete with Time/Ripple

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 548)".

Wave Projects

In "Wave editing" mode, only those wave project segments below the selected area remain. The files before and behind the range are 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

This command cuts the objects apart at the position of the silence. Before applying the command, add a threshold value in the dialog for the silence and used signal. The cut objects are marked 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 enables you to 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.

Clear

The data of the selected area will be deleted. The files found after the selected area will not be moved, leaving silence in the tracks selected. The length of the track stays unchanged.

Shortcut: Alt + Del

Tempo

Tempo Map Dialog...

Detailed information about the "Tempo Map" dialog is available in chapter "Tempo Editing (view page 300)".

Tempo Map BPM Mode

Detailed information about the "Tempo Map BPM Mode" is available in chapter "Tempo Editing > Tempo Map Modes: BPM Mode / Grid Fit Mode (view page 302)".

Tempo Map Grid Fit Mode

Detailed information about the "Tempo Map Grid Fit Mode" is available in chapter "Tempo Editing > Grid Tapping" (view page 305).

Insert Tempo Change

Here you open up the "Tempo and Time Signature Dialog (view page 310)" in order to insert tempo change.

Insert Time Signature Change

Here you open up the "Tempo and Time Signature Dialog (view page 310)" in order to insert time signature change.

Insert Grid Position Marker

Detailed information about the "Grid Position Marker" is available in chapter "Tempo Editing > Grid Position Marker". (view page 305)

Tempo Scaling

With this menu command all BPM values of the selected tempo curve points can be scaled with one adjustable factor.

Ignore All Tempo Markers / Use Project Tempo

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 84)".

Create click track

The new command **Menu Edit > Tempo > Create click track** creates an audio track, which contains all metronome click 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 Right

This function moves the beginning of the range in the current arranger window by one grid unit to the right. 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 End Left

This function moves the end 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), "Shift + right arrow"

Move Range End Right

This function moves the end of the range in the current arranger window one grid unit to the right. 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); 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.

Range Over All Selected Objects

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 play cursor 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 149)".

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

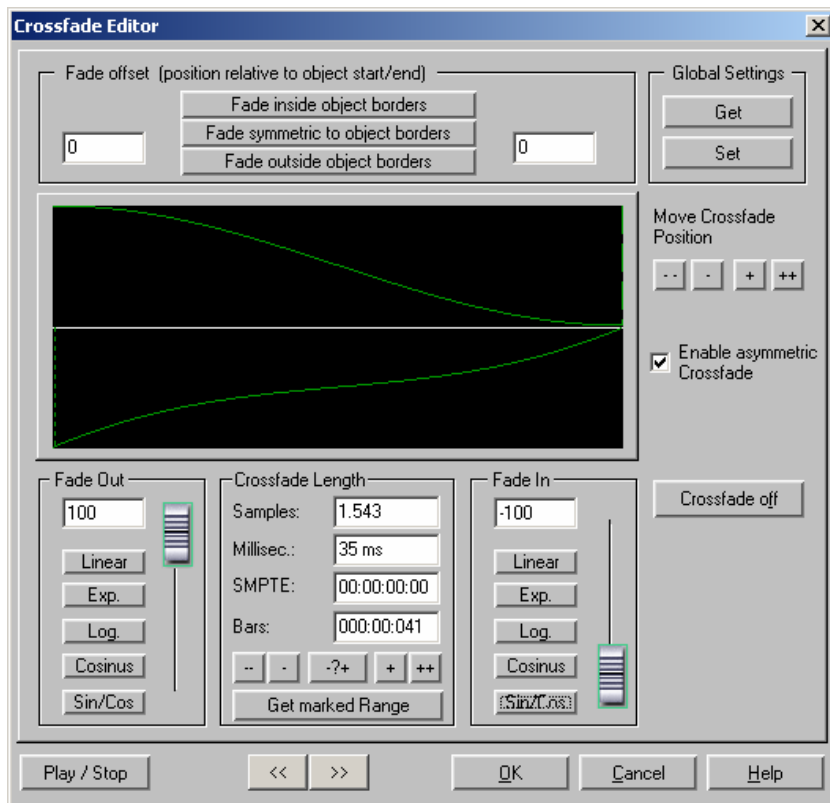
Samplitude features a professional crossfade editor.

The crossfade editor is opened by clicking the toolbar button or via menu "Edit -> Crossfade editor". You can also use the keyboard shortcut "Ctrl + 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.

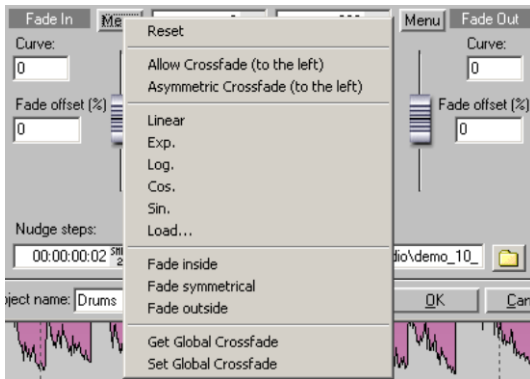
Move crossfade position: The end of the first object and the end of the second object are moved equally. The ++/--/+/- buttons are used to set the crossfade length.

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 Active

With this option, all newly recorded or cut material in a VIP (or material that is dragged from a wave project into a VIP track) 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 may manually edit each intersection with the crossfade editor or handles.

Batch Processing...

Use batch-processing to automate long work processes. This makes it possible to define an editing sequence, which will be applied to all selected files.

Every task will be entered into the task list. Define multiple batch tasks, which will then be executed one after the other.

Possible editing options:

- Normalization
- Linear fading (in and out)
- Real-time effects
- Time Stretching/Pitch Shifting
- De-clipping
- Remove DC offset
- Target format settings: Word width (8/16/24/32 float), sample rate, stereo/mono settings, file format conversion, compression process,
- memory settings.

For example, you can normalize a whole list 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

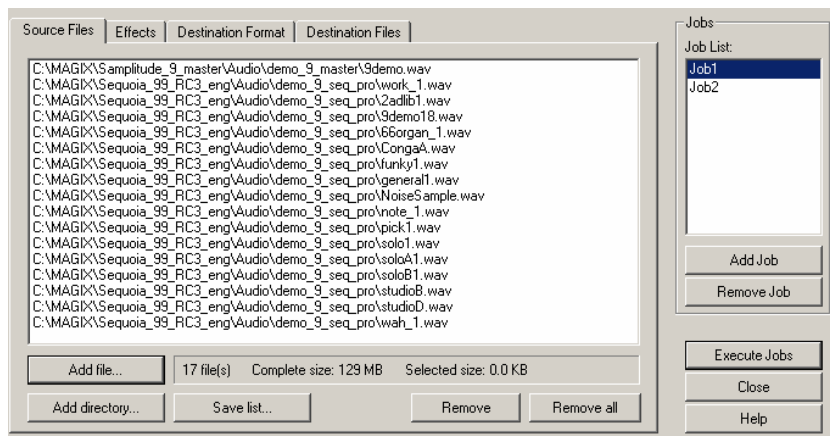
compressor, change them all to 16-bit mono, resample them on 22 kHz, and then save them all as MP3 files.

Tasks

The task list is on the right edge of the batch processing window. New jobs are created by activating 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). **"Remove job"** removes the selected job. **"Perform jobs"** starts the batch processing.

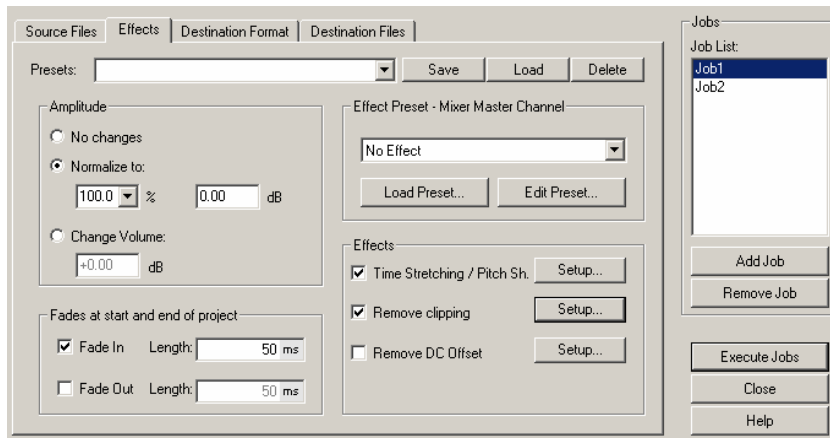
As long you don't close down Samplitude, all the tasks remain saved until execution, even if you close the batch processing window in the meantime.

Source Files



Create the list of the files for editing with the **"Add file"** button. Multiple selection is possible. Select **"Add directory"** to add all of the audio files within a directory (including all subfolders). All file formats that Samplitude can import may be loaded. **"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: Enter a target maximum amplitude value in % or dB. For example, a value of 75% corresponds to -2.5. Alternatively, you may also enter volume changes in dB. Detailed information about normalization is available in "Effects -> Amplitude -> Normalize (view page 603)".

2. Master effects: To avoid redundancy in the batch processing dialog, effects settings have been integrated with the mixer preset files.

The Samplitude mixer enables all of the mixer settings to be exported as a mixer preset. Click the "Save mixer settings" button in the mixer. This is located under the snapshot storage slots.

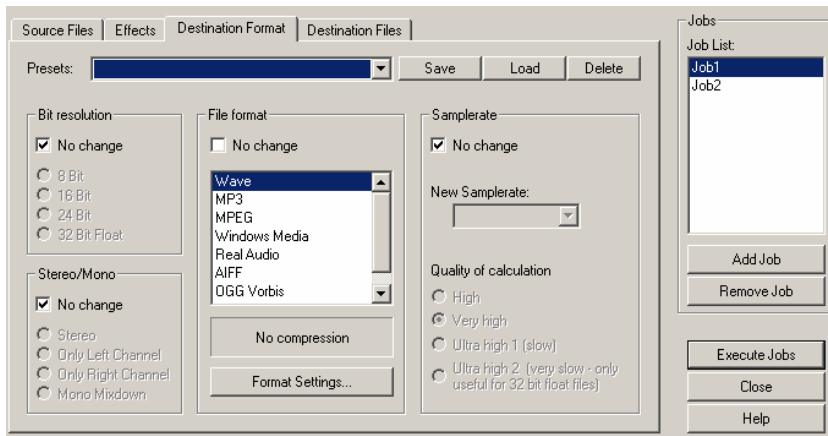
When you load the *.mix file into the batch-processing function via "**Load preset**", all effect settings for the master channel will be calculated into the file batch. "**Edit preset**" opens a special FX routing dialog that provides access to all of the effect parameters.

Detailed information about the FX routing dialog is provided in the chapter "Mixer - buses and routing -> Effects routing/plugin-ins dialog (view page 174)".

Additional effects: Besides the master effects of the mixer presets, non-real-time effects such as "Resampling/Time Stretching", "Remove DC offset", and "Remove modulation" may also be applied. The "Settings" button opens the corresponding effects dialog.

3. Fades at the start and end of projects Linear fades may be added to the start and end of files. The fade lengths may be set freely.

Target format

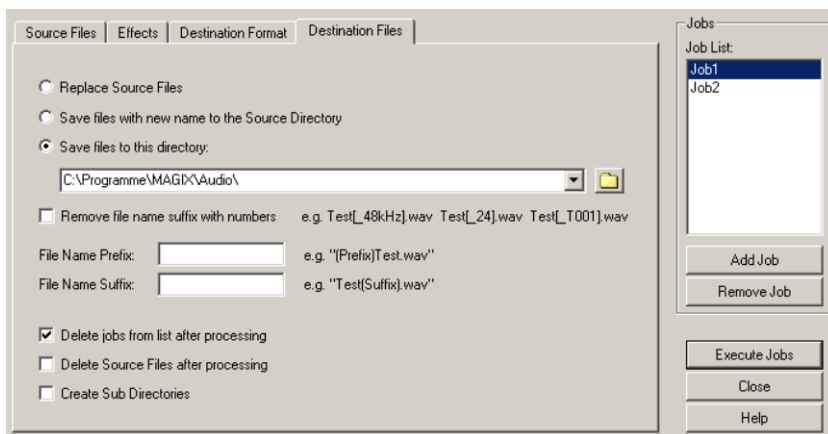


All settings in this dialog can be saved as presets. You can freely choose the bit resolution (8/16/24/32 bit float), the sample rate, stereo/mono/left/right as well as the save format with the corresponding format settings.

You can find out more about sample rates in the "File > Effects > Sample manipulation > Adjust sample Rate (view page 682)".

You can find out more about file formats in the "Menu > File > Export (view page 449)".

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 will be closed first.

Save files with new name to the source directory: The edited file is copied back into the source directory and the file name will be expanded with an additional "**file name prefix**" and "**file name suffix**".

Save file in the following directory: The edited file will be saved in the selected directory; the file name will be extended to include the "**file name prefix**" and "**file name suffix**".

Remove file name suffix with numbers: This option removes numbers added by Samplitude's 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**" may also be applied.

Create subdirectories: This option saves all files in the same subdirectory structure including the source path. Subdirectories are created as required.

More

Range Editor...

This window specifies the currently marked range's start and end positions and length in various measurement units.

Range Start	Range Length	Range End
Samples: 26.460	Samples: 52.920	Samples: 79.380
Millisec.: 600 ms	Millisec.: 1.200 ms	Millisec.: 1.800 ms
SMPTE: 00:00:00:15	SMPTE: 00:00:01:05	SMPTE: 00:00:01:20
Bar: 001:02:000	Bar: 000:02:000	Bar: 001:04:000

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 Tracks...

To add multiple tracks, select the command "Insert several tracks". A dialog window appears to select the desired number of new tracks.

New MIDI Track

This command adds a MIDI track to the end of a selected track.

New Folder Track

This command adds a folder track in front of the selected track. If a range across several tracks is selected, each of these tracks will be added to the new folder track.

Detailed information about folder tracks is available in "Screen elements -> Program interface - overview -> Folder tracks (view page 61)".

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 426) as a track here. The new bus will be added below the selected track.

New Surround Bus

You can mix down your project in Surround sound retroactively. Use this command to create a Surround bus that will be placed below the selected track. All tracks routed to one Surround bus receive a Surround panorama module instead of the normal panorama slider to control the Surround position of the output signal.

As soon as you have created a Surround bus, you can route the exiting signal of an object to this Surround bus and set the Surround panorama independently from the track panorama settings.

The Surround master is created simultaneously when a Surround bus is created; the individual channels are routed to the different output devices.

New Surround AUX Bus

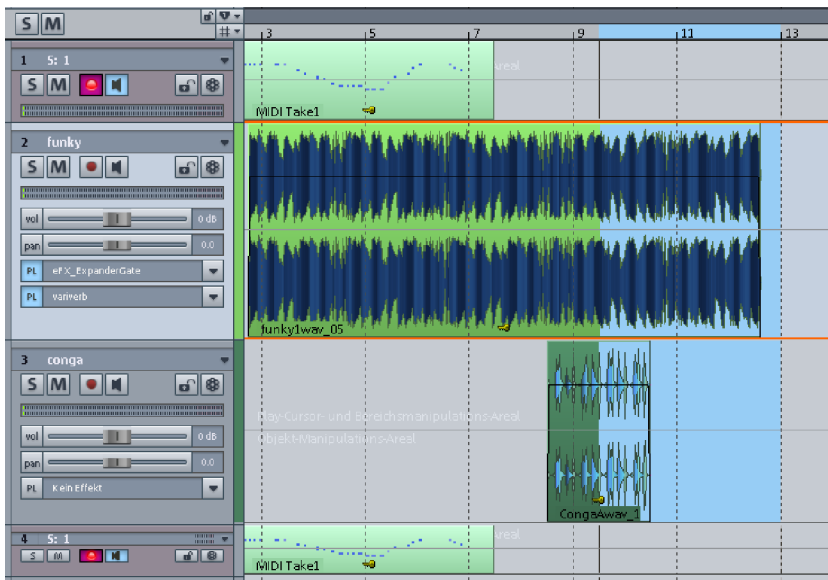
Use this command to create a new Surround AUX Bus.

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 tracks

Use this function to copy tracks that are selected to the track header (view page 98), i.e. in the clipboard.

If you want to copy several sequential tracks, drag out a selection over all of the tracks that should be copied. The selection does not need to be adjusted to the track contents.



Next, load the selected tracks into the clipboard via "Copy tracks", and then insert them as additional tracks above the last selected track with the command "Insert tracks". This results in all track settings, save the AUX send paths, being adopted.

Paste Track(s)

This function inserts all tracks copied into the clipboard (VirtClip (view page 427)) above the selected track.

Deletes 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 28)

Input

Stereo in: The channel input is switched to stereo and the signal will be picked up on two channels.

Mono in: The channel input is switched to mono and the signal will only be picked up on one channel.

Mono mix in: The track input is switched to mono, the signal is recorded like a stereo recording (dual channels), and then mixed into a single channel.

MIDI track record: If you click this menu entry the MIDI section of the track editor will open, allowing you to record MIDI files.

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 hides the selected track from being displayed in the track view. It will, however, remain visible in the mixer and will continue to play there.

The menu item "Show all tracks" shows this and all other hidden tracks.

An overview of all tracks including their status of visibility is included in the track manager. ("Tools -> Manager -> Track manager" shortcut: Ctrl + Shift + S). Hidden tracks may 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 Manager -> Track manager" section.

Unhide All Tracks

Use this command to show all hidden tracks in the arranger once again.

Track size

Maximize Track

The activated track is maximized, i.e. it is displayed in a large view.

Minimize None

All minimized (and maximized tracks) are displayed in the same size.

Display Subtracks

This command opens the subtracks for a folder track.

Freeze Track

With "Track freeze", you can have the selected file rendered into a wave file which replaces every object on the chosen track. The object and track effects are also added, reducing 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 effectively, highly flexibly, and in a way that reduces strain on the computer's resources.

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.

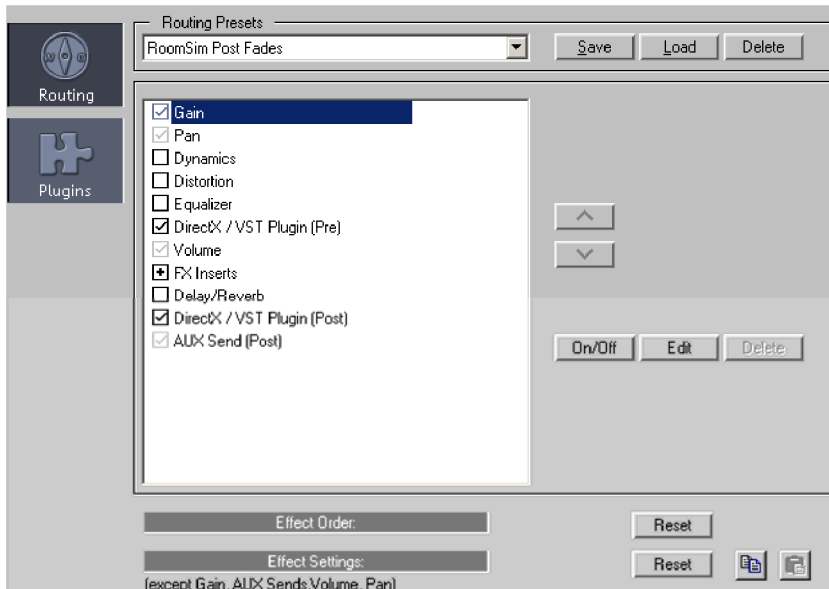
Changes made in the temporary freeze project are also applied to the track.

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

Track Routing Dialog

You can apply all important settings for real-time effects in the track routing dialog.



Track 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 respective effect. The check boxes for volume, pan and gain knobs as well as AUX gain knobs are grayed-out, because they are represented in the list only for the purpose of ordering the effects.

The order of the effects in the mixer tracks and the mixer master may be configured freely; however, in the object, this order is restricted. You can change the position of an effect with the up/down buttons in the "Effects sequence" section. "Reset" changes the order back to the previous sequence.

There are two entries for "AUX send" (pre and post) in the mixer tracks. 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 junction off the AUX signal to the AUX bus. DirectX and VST plug-ins may also be routed pre and post in a similarly flexible manner.

Track Routing Dialog - Dialog Buttons

Save/Load/Delete: Manage your individually customized settings here.

All Tracks: The respective settings are applied to all tracks.

Reset: Returns all settings to the Samplitude basic settings.

Copy/Paste: Use these buttons to quickly apply the settings to different tracks/objects.

Edit: Opens the dialog for the active effect/plug-in. This can also be achieved by right clicking the desired list entry.

On/Off: Switches the respective activated effect "on" or "off". Activating or deactivating the check box preceding the corresponding effect/plug-in has the same effect.

Remove: You can also remove internal Samplitude effects (FX inserts) instead of just deactivating them if you want to save resources. "Unloaded" effects are displayed with a minus sign in the box on the left side.

Play/Stop: Plays and stops playback at the current cursor position.

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 DirectX and 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 DirectX and 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.

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.

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...

The screenshot shows the 'Track Options' dialog box with three main sections: Audio, MIDI, and Record.

- Audio Section:**
 - Record: M-Audio Delta ASIO In (1+2) [All Tracks] [Track/Dev +]
 - Output: M-Audio Delta ASIO Out (1+2) [All Tracks] [Track/Dev +]
 - Track Type: Submix Bus AUX Bus
 - Economy: 2 Channel Surround active:
 - Time Shift / Delay: 0 Smp [Effects/Routing]
 - Color: [Blue] [Reset]
- MIDI Section:**
 - Record Dev.: Creative Prodigies [All Tracks] [VST Instrument Editor]
 - Playback Dev.: VSTi 2: Revolva 2 [All Tracks]
 - Record controller curves while Playing [Automation/Controller ...]
 - Playback/Thru parameters:

In Chan.	Out Chan.	Transp.	Program	Bank: High	Low
all	all	0	no change		
 - MIDI Thru active
- Record Section:**
 - File: WS407.wav [Folder Icon] [All Tracks...]
 - REC [Audio] [MIDI]
 - Audio Options: Stereo Record only to RAM
 - Mono Mono (Mix)

At the bottom, there is a 'Max' checkbox, a 'Track name' field containing 'Bass', and navigation buttons '<<' '>>', 'OK', 'Cancel', and 'Help'.

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 28)" 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 play cursor display to be delayed by the amount entered. Negative values cause playback to occur earlier. All other tracks and the play cursor will be delayed in relation to the entered value. To the right of the input field, you can select the unit for the time delay.

Colour: 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.

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.

MIDI

The "MIDI" section provides access to the various MIDI options.

Recording Device (MIDI): Use this menu to choose a MIDI entry device.

All Tracks: Applies the same recording device selected for this track to all tracks.

VST Instrument Editor: This button opens the graphical interface for the connected VST instruments.

Playback Device (MIDI): Use this menu to select the MIDI playback device. The VST instruments may also be selected as playback devices.

All Tracks: Sets the same recording and playback device as selected for this track for all tracks.

Record Controller Curves During Playback: This option enables controller curves to be recorded during playback.

The following are available as additional playback parameters: in-channel, out-channel, transpose, program change, bank high, bank low. You can also activate **MIDI Thru** for the affected track. This activates the loudspeaker symbol for this track.

Automation/Controller: This button opens a dialog to change the settings of the MIDI controller.

Detailed information about controller automation is provided in "Automation -> MIDI controller automation (view page 338)".

Track Name: Use this field to change the name of the current track.

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

Cut tracks

This function cuts out all tracks that are selected in the track header (view page 98). 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 178)".

- AUX sends: Opens the AUX send routing dialog window
- Dynamics: Dynamics, Advanced dynamics, Multiband dynamics, sMax11, efx_Compressor, eFX_Gate, AM-Munition, AM-Track, AM-Phibia, AM-Pulse
- Frequency/filters: Parametric EQ, EQ116, FFT filter, brilliance enhancer, Filtox, eFX_DeEsser
- Delay/reverb: Delay, room simulator, efx_Reverb, efx_StereoDelay, Ecox, Variverb
- Distortion: Distortion, Ampsimulation, Vandal, eFX_VocalStrip,
- Restoration: DeClipper, DeClicker/DeCrackler, DeHisser, DeNoiser
- Stereo/phase: Pan/Surround editor, Multiband Stereo Enhancer, track phase inversion
- Modulation/special: Vocoder, efx_ChorusFlanger, efx_Phaser, eFX_TremoloPan, Corvex
- MIDI: Velocity Dynamics
- Plug-ins: opens the VST plug-in dialog. Here you can select MAGIX plug-ins, VST effect plug-ins as well as DirectX plug-ins from 3rd parties from the "**Add Plug-ins**" dialog, you can also call up the "VST/Direct X/Rewire Setup".
- essentialFX: eFX_ChorusFlanger, eFX_Phaser, eFX_Reverb, eFX_StereoDelay, eFX_Compressor, eFX_Gate, eFX_DeEsser, eFX_VocalStrip, eFX_TubeStage, eFX_TremoloPan
- MAGIX plug-ins: AM-Munition, AM-Track, AM-Phibia, AM-Pulse, Corvex, Ecox, Filtox, VariVerb, Vandal

Track properties

This submenu enables access to all track properties.

Mute: Mutes the active track.

Shortcut: Alt + M

Mute inactive: Mutes and also deactivates the track. This increases performance as the caching and FX processing of this track are not necessary)

Shortcut: Ctrl + Alt + M

Solo: Switches the active track to "Solo" mode.

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 are silenced in solo mode.

Shortcut: Shift+ Alt + S

Record: Arms the track for recording.

Shortcut: Alt + R

Monitoring: Switches manual monitoring for the corresponding track on/off.

Shortcut: Ctrl + Alt + Shift + F

Lock: Locks the active track.

Shortcut: Alt + L

Edit volume/Panorama: Enter a numerical value for volume in dB or for the panorama curve.

Shortcut: Ctrl + Shift + K; Ctrl + Shift + P

Track phase inversion: Rotates the phase of the selected track.

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: Enable/disable all mute states of the tracks.

Global solo: Enable/disable all solo states of the tracks.

Global: solo safe: All tracks will be switched to "Solo safe" mode. In this mode the corresponding track, if switched to solo, will be monitored together with its AUX returns.

Global: solo PFL: All tracks will be switched to "Solo PFL" mode. In this mode the corresponding track, if switched to solo, will be tapped at a position before the fader and heard as a pre-fader signal.

Global: solo exclusive: All tracks will be switched to "Solo exclusive" mode. In this mode, the corresponding track will play exclusively if switched to solo. Solo tracks already switched to solo mode will be switched back.

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 as well as MAGIX synths (e.g. Robota Pro) may be integrated seamlessly into a virtual project and controlled via 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 produce results, this means that a VST instrument has not been loaded into the corresponding track.

Detailed information about plug-ins can be found in "Software/VST instruments (view page 312)".

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 114) 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 247)".

Edit audio file

Here the wave project belonging to an object opens and is displayed in the wave window. The range corresponds precisely to the part of the audio data which the object accesses.

Detailed information about offline audio editing and real time is available in "Working in the project window -> Samplitude as a wave editor (view page 108)".

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

A 4-beat loop object is created at the current play cursor position. The instrument interface for the synth object that is created may be accessed by double clicking the object. The synth objects are Atmos, BeatBox 2, DrumnBass, and LIViD.

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 cursor. Note that 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.

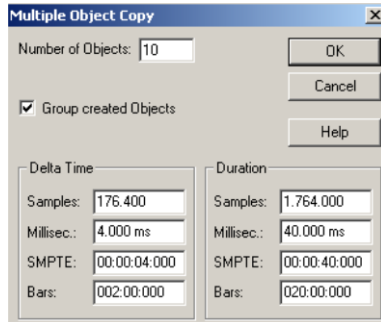
Duplicate Objects Multiple

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 Marker Positions

The object is split into pieces at the marker positions that lie within the object borders. The newly created objects will be named after the previous markers.

Split Objects at Track Marker Positions

The object is split into pieces at the marker positions that lie within the object borders. The newly created objects will be named after the previous track markers.

Split Objects at Audio Marker Positions

The object is split into pieces at the audio marker positions that lie 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...".

Edit Copy of the Wave Project

Use this command to create a copy of the wave project featuring the object first. This copy may then be edited using the "Wave editing" function.

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 456)

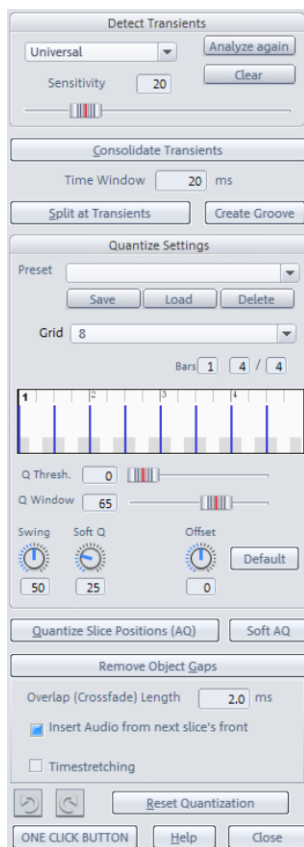
and are now visible in the arranger window of the new project. It's possible that you may want to change something else in the song retroactively. "Edit root VIP" opens the original project to make the desired changes. Once the modified project is saved, it will be rebounded and updated in the project that is used to compile the CD.

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.

Quantization

Audio Quantization Wizard



The quantization wizard also lets you apply the functions described above for transients and beat marker editing. These commands reflect the typical work process when adjusting a multi-track drum recording to the VIP grid and are applied to object selection. The **"One click button"** automatically executes the **"Determine transients"**, **"Consolidate transients"**, **"Cut at transients"**, **"Quantize object positions (AQ)"**, and **"Remove gaps between objects"** 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.

Use the **"Analyze again"** and **"Clear"** buttons to 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"**.

Individual gaps between objects may be closed either by Object Time Stretching or by using the audio material from the slices to the right beside the gap.

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 marker Manager (view page 589)".

Consolidate Transients

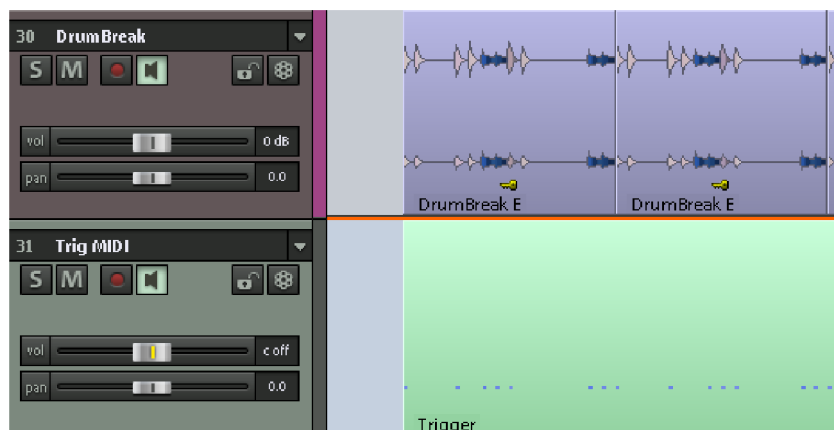
Transients within a certain range (e.g. 20 ms) are moved to the first transient position with this command. The effect is that "Split at transients" doesn't create too many splices, especially if you have several tracks with slightly different transient positions for the same drum hit, i.e. due to microphone 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 552)" in the Quantization wizard.

Quantization setting - 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 Audio Object Quantization

The "Reset object 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

Quantize Settings - Grid

Use the "Grid" parameter to set the selection step length and thereby the values for quantization. A value of "1" provides a snap value of a single note, "2" is a half note, "4" is a quarter note, etc. There is also a series of decimal values, n-tuplets, and a selection of special groove templates.

Detailed information about groove templates is available in "MIDI editors -> Quantize -> Quantize settings - Groove template (view page 260)"

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 $\frac{1}{4}$ of the grid width left and right of the grid point ($\frac{1}{16}$ 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.

Audio Quantization – Tutorial

1. The first step is to enable display of **audio markers/transients** in the display options ("Y -> Design -> Display options -> Options -> Audio markers/transients").
2. Load a VIP with a multi-track drum recording where the drummer has played to the click, so that the VIP raster matches the drumming tempo.
3. Select a range for audio quantization. We recommend around 8 - 16 beats to guarantee fast calculation. Analysis of transients for a complete song can take a little longer. Next, separate the objects into the selected range size.
4. Select all drum objects, open the audio quantization dialog, and move the **"Sensitivity"** slider. Beat markers should appear within the object at the transient positions. Adjust the sensitivity so all main hits are detected. Use the "F2" and "F3" keys to jump 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. Amplitude 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, access the **"Split at transients"** command.
7. Use **"Quantize object positions"** to adjust the individual positions of the partial objects. You also have the option of changing the quantization settings.
8. Audio gaps might appear between some slices due to quantization. These can be filled with the option **"Remove gaps between objects"**.

If you are not satisfied with the quantization result, you can **"Reset quantization"** to set the split objects back to their original position and try other quantization settings.

When you have finished editing, you can **Glue slice objects together** (shortcut: Ctrl + Alt + G) on each track to save CPU performance.

MIDI Note Quantize (Standard)

This command quantizes the ends of the MIDI notes of all selected MIDI objects according to the MIDI quantization settings.

Detailed information about quantization is available in "MIDI editors -> Quantization".

MIDI Humanize Q

This command applies the current **Humanize** value in the quantization options.

Detailed information about quantization is available in "MIDI editors -> Quantization".

MIDI Quantize Settings...

Use this command to open the dialog for quantizing MIDI events.

Detailed information about quantization is available in "MIDI editors -> Quantization".

Extended MIDI Quantize

Start Q

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 set start quantization value.

Detailed information about quantization is available in "MIDI editors -> Quantization".

Start and Length Q

This command applies quantization to the start and length of the MIDI notes of all selected MIDI objects according to the MIDI quantization settings.

Detailed information about quantization is available in "MIDI editors -> Quantization".

Length Q

This command applies quantization to the lengths of the MIDI notes of all selected MIDI objects according to the MIDI quantization settings.

Detailed information about quantization is available in "MIDI editors -> Quantization".

Soft Q (approximate quantization)

This command applies the current Soft Q (level) value in the quantization options.

Detailed information about quantization is available in "MIDI editors -> Quantization".

Quantize Note Ends to Grid

This command quantizes the ends of the MIDI notes of all selected MIDI objects according to the MIDI quantization settings.

Detailed information about quantization is available in "MIDI editors -> 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 is available in "MIDI editors -> Quantization".

MIDI Input Q (global)

Use this option to quantize the MIDI recording destructively. The original position may be restored via the menu item "Object -> Quantize -> Reset Quantization".

Object effects

The following effects may also be accessed at the object level.

Dynamics: Dynamics, advanced dynamics, multiband dynamics, sMax11, efx_Compressor, eFX_Gate, am-munition, am-track, am-phia, am-pulse

Frequency/Filter: Parametric EQ, EQ 116, FFT-Filter/Spectral analysis, Brilliance Enhancer (optional), Filtox, eFX_DeEsser.

Delay/Reverb: Delay, RoomSimulator, eFX_Reverb, eFX_StereoDelay, Ecox, VariVerb.

Time/Pitch: Resampling/Timestretching, Elastic Audio.

Distortion: Distortion, Vandal, eFX_VocalStrip, eFX_TubeStage.

Restoration: DeClipper (optional), DeClicker/DeCrackler (optional), DeHisser, DeNoiser (optional).

Stereo/phase: Object channels, Switch channels, multiband stereo enhancer, invert phase (both channels, left channel, right channel)

Modulation/special: Vocoder, efx_ChorusFlanger, efx_Phaser, eFX_TremoloPan, Corvex, object reverse

Plug-ins: opens the VST plug-in dialog. Here you can select MAGIX plug-ins, VST effect plug-ins as well as DirectX plug-ins from 3rd parties from the "**Add Plug-ins**" dialog, you can also call up the "VST/Direct X/Rewire Setup".

essentialFX: eFX_ChorusFlanger, eFX_Phaser, eFX_Reverb, eFX_StereoDelay, eFX_Compressor, eFX_Gate, eFX_DeEsser, eFX_VocalStrip, eFX_TubeStage, eFX_TremoloPan.

MAGIX Plug-ins: AM-Track, AM-Phibia, AM-Pulse, AM-Munition, Corvex, Ecox, Filtox, VariVerb, Vandal.

Selecting Objects

Select All Objects Under Cursor

Selects all objects found under the play cursor 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 Previous Object

Use this command to add the previous object on the same track to the current selection.

Shortcut: Ctrl + Alt + Shift + Q

Select 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

Deselects all objects.

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 133)"

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

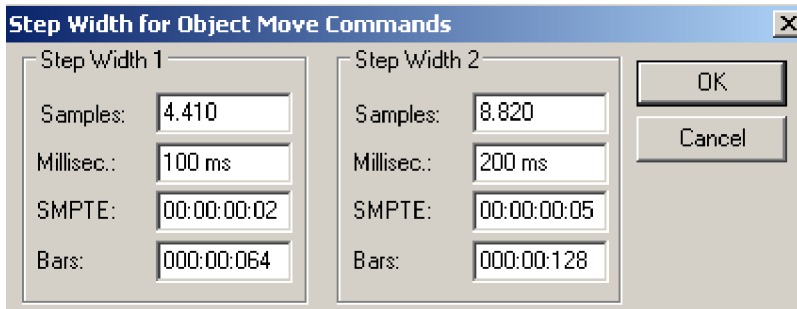
Use this function to edit the numerical interval between the selected object and the object in front of it on the same track. With multi object selection the entire distance between all objects on a track can be set.

Object/Fade step size

Select an object for simple object editing.

If you have selected two objects for editing, it is possible to edit 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. Doing this 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. Moreover press a key on the num pad 0-9, and 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
Move object(s) left	Ctrl + Alt + 1
Move object(s) 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
Move crossfade left	Ctrl + Alt + 3
Move crossfade right	Ctrl + Alt + 4
Start offset to the left	Ctrl + 5
Start offset to the right	Ctrl + 6
End offset to the left	Alt + 5
End offset to the right	Alt + 6
Increase left volume	Ctrl + 8
Reduce left volume	Ctrl + 7
Increase right volume	Alt + 8
Reduce right volume	Alt + 7
Increase volume	Ctrl + Alt + 8
Reduce volume	Ctrl + Alt + 7
Left object content to the left	Ctrl + 9
Left object content to the right	Ctrl + 0
Right object content to the left	Alt + 9
Right object content to the right	Alt + 0
Object(s) content to the left	Ctrl + Alt + 9
Move Object(s) content right	Ctrl + Alt + 0
Additional key for step size 2	Shift key

Please note the useful "Object content to the left/right" functions. Here, object length and object position are kept, the assigned audio material from the corresponding wave project is moved into it.

Object Hotspot to Play Cursor Position

Use this command to move the hotspot of the selected object to the play cursor 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 play cursor 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 play cursor 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 Original Time Position

This command opens a dialog which lets you write a new original position into the broadcast WAV header. The current object position is used as the new position, which may then be changed manually.

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 play cursor 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 play cursor by using the function "Object hotspot to play cursor position (Shortcut Ctrl + Alt+ P)" (view page 561).

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 150)".

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 152)".

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 "Comparisons colors" for the waveform color in the display options ("View -> VIP display mode -> Definition (view page 491)"), 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".

Freeze Objects

This function **renders each selected object as a new wave file**. The original object is replaced in the arranger. Freezing increases performance in case a very system-intensive real-time plug-in is active. The original object is maintained at all times and may be re-edited using the "Edit object freeze" function or put back into the arranger via "Unfreeze objects".

"**Freeze MIDI object**" replaces the MIDI objects in the audio return signal of a software instrument featuring audio objects. The audio return signal must be routed to the MIDI track to do this.

If more than one object is selected, the 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 multi-track 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

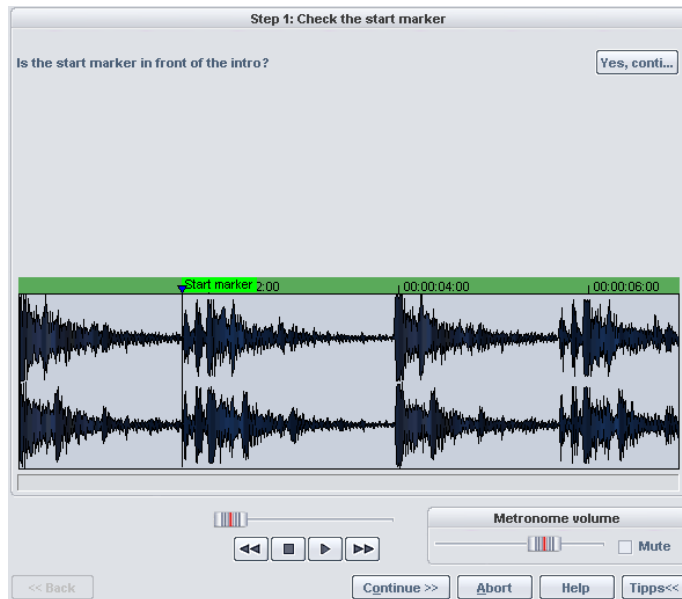
Start the Remix Agent from the "Tools" menu or from the context menu by right clicking the object.

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 four 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 play cursor 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 play cursor 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

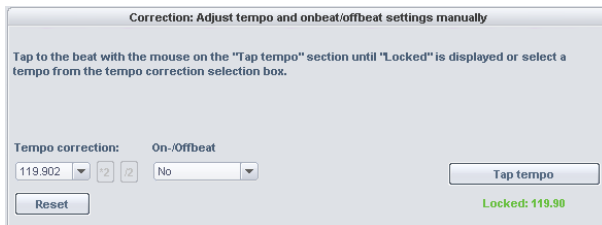
After opening the Remix Agent by pressing the "Continue" button, it will begin to analyze the audio material and try to determine the tempo. The object is then played with the result sounding like a metronome click and numbered green beat lines appear in the waveform display.

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.

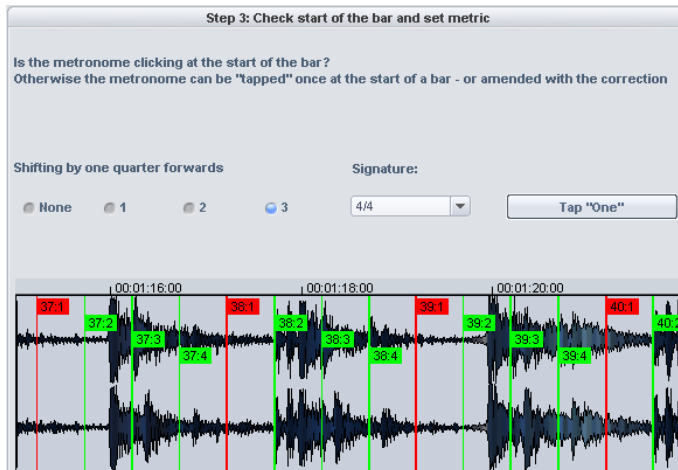
Use the "0" key to manually set the quarter beats while the music plays. Stray markers are automatically removed in so that the set tempo remains fundamentally intact.

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 bar 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.

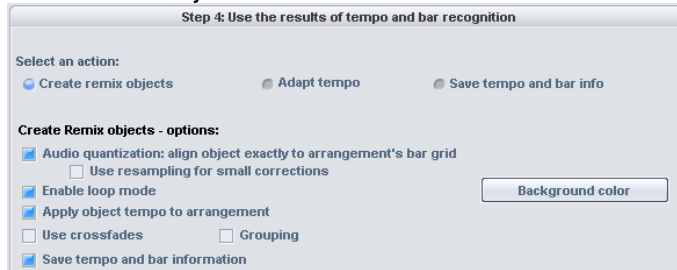
Continue to the last step once the starts of the bars are now correct.

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 multi-track 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.

The screenshot shows a settings panel with the following options:

- Select an action:**
 - Create remix objects
 - Adapt tempo
 - Save tempo and bar info
- Tempo alignment - options:**
 - Apply arrangement tempo to object tempo
 - Apply object tempo to arrangement tempo
 - Use time stretching
 - Use resampling
 - Use audio quantization
 - Save tempo and bar information
 - Change global settings
 - Adapt tempo map
 - Use quarters
 - Use bar positions instead of tempo map

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 multi-track 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.

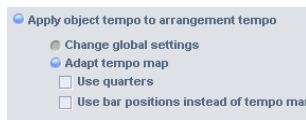
Save tempo & bar info into the audio file: If this option is active, the tempo and bar information are saved in the audio file. The objects in the virtual project won't be changed.

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 play cursor 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.

audioid

Detailed information on audioid can be found in "Menu CD/DVD > audioid... (view page 795)".

Comparisons Audio Search...

This function uses the patented Comparisons 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 an HD wave project. Now select the range to examine and copy it into the clipboard (shortcut: "C").

Now start the audio comparison function.

The dialog that appears enables the sensitivity threshold for the recognition algorithm to be changed. During the adjustment process, markers appear in the project specified as "Match markers" at positions where the algorithm detects similar audio passages.

After closing the search dialog, you may jump to the ranges that were detected by clicking the left border of the corresponding "match" marker, even during playback.

Time Stretch/Pitch Shift Patcher...

Timestretching / Pitchshifting / Resampling

Algorithm	SD	BB	ET	TL	OR	SP	MG	EG	MIX	BM	CPU%	TS factor
Resample	+++	+++	+++	+++	+++	+++	+++	+++	+++	+++	+++ / 22.6	0.75 - 4.0
Standard	+	+	+	+	+	+	+	+	+	+	2.0	0.80 - 3.2
EQ stretching (Certs)	++	++	+	+	+	+	+	+	+	+	1.0	0.50 - 2.0
Smoothed	-	-	+	++	+++	++	+	+	+	+	-	0.50 - 2.0
EQ stretching	+++	+++	+++	+++	+++	+++	+++	+++	+++	+++	1.0	0.25 - 4.0
EQ stretching	++	++	++	++	++	++	++	++	++	++	1.0	0.80 - 3.2
Monophonic voice	++	++	+++	+++	+++	+++	+++	+++	+++	+++	1.0	0.70 - 3.5
Universal HD	++	++	+++	+++	+++	+++	+++	+++	+++	+++	1.0	0.70 - 3.5

SD - Complex down loops, **BB** - Disableable down loops, **MG** - Polyphonic instruments, **ET** - Monophonic instruments, **TL** - Surface, **OR** - Orchestra
SP - Speech, **MG** - Polyphonic vocals, **EG** - Monophonic vocals
BM - Beat marker (list)
 - Quality/resampling evaluation
 - Exact synchronization to beat marker position
 - ++ = low load, +++ = high load
 - RS: resampling with constant regular (DR) load vs. standard resampling
 - Recommended range for timestretch factor

Universal HD
 Universal method for timestretching and pitchshifting in very high audio quality.
 Beat markers are evaluated to improve audio quality.
 Suitable for all types of audio material.
 This method requires so much processing time that a real-time application is recommended only in exceptional situations. Using the bounce function is recommended instead.

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)

Time Stretch/Pitch Shift Patcher – Patching

Patches only work if you have opened the wave project in destructive "Wave editing" mode and if it is not open in a VIP simultaneously.

Choose the algorithm you wish to use in the dialog. You may also specify the tempo of the wave file using the "Time factor" controller.

You may now test the selected algorithm in conjunction with the "Play/stop" and "Play original" buttons. By clicking the "Patch" button, Samplitude writes the information permanently to the wave file.

Algorithms for Time Stretching/Pitch Shifting

Samplitude features the "Beat marker stretching (smoothed)", "Smoothed", "Beat marker slicing", "Beat marker stretching" as well as "Monophonic voice" algorithms as well as the "Standard" algorithm for real-time Time Stretching.

Detailed information about this is available in "Offline effects -> Resampling/Time Stretching/Pitch Shifting (view page 647)". Use the patcher to save the most suitable time stretch algorithm for your needs in the wave project so that the chosen algorithm is used automatically on your wave projects when using Time Stretching/Pitch Shifting.

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 517), 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.

Object Manager...

The object manager displays all of the objects used in the arranger window in chronological order.

If you select objects by ticking the box in the "Track" column, they will be selected in the arranger as well. This makes the selection of objects a speedy process.

Detailed information about the object manager is available in the chapter "Managers -> Object manager (view page 143)".

Shortcut: Ctrl + Shift + O

Playback 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 52) as well as the "Play once" button in the toolbar (view page 66).

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 52) as well as the "Play loop" button in the toolbar (view page 66).

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 (view page 66).

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 cursor is not already 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 503)".

Shortcut: F5

From play to cut start (in point)

A defined time segment from the start of the selected range is played. The duration of this segment may be set under "Options -> Program preferences -> Set pre-roll time (view page 503)".

Shortcut: F6

Play to cut end (out point)

A defined time segment is played until the end of the selected range. The duration of this segment may be set under "Options -> Program preferences -> Set pre-roll time (view page 503)".

Shortcut: F7

Play from cut end (out point)

A defined time segment from the end of the selected range is played. The duration of this segment may be set under "Options -> Program preferences -> Set pre-roll time (view page 503)".

Shortcut: F8

Play over cut/crossfade

This allows you to simulate the cut, in which case a defined segment is played until the beginning of the selected range. The selected range is then skipped and a defined segment is played from the end of the selected range. The duration of this segment may be set under "Options -> Program preferences -> Set pre-roll time (view page 503)".

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 play cursor 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 play cursor 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 play cursor will remain at the current position.

Shortcut: Number block "0" or ", "

Playback Mode

Loop Mode

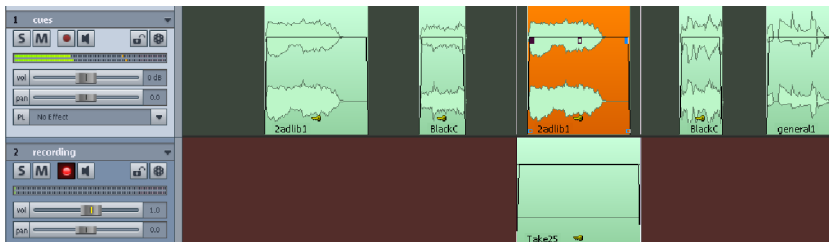
This mode plays the selected ranges in a loop. This corresponds to switching on the "Loop" button in the transport control.

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 playable material like original sounds in radio broadcasts and theater. The objects contained in the original sound track (= cues) may be played back in sequence while they are recorded in another parallel track.



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 etc. Related objects are treated as an object.

Of course, multi-track projects may be played in cue mode. The object on the first track will be used for the Start/Stop range markings.

2. Control via 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 so, switch on "CD arrangement" mode in the "CD/DVD" menu. 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 point for the next playback range.

Note: For smooth Cue mode use deactivate the Loop mode.

In cue mode, the recording mode in the transport control changes to **Record without playback (band control)** (view page 581). The original sound 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 play cursor may be repositioned at any time within the currently played original sound.

"System options -> Playback (view page 477)" may be accessed to ensure that the **first track in cue mode is switched to "Solo"**. **Pre-roll (ms)** may be used to specify in milliseconds how long the delay between execution of the record command and the actual start of the recording should be.

Scrubbing Active

- Scrubbing Active Keyboard shortcut: "**Alt + Shift + Page down**"
- Jog (Absolute)
- Two Speed
- Shuttle (Relative)
- Scrub Left Keyboard shortcut: "**Alt + Shift + Arrow left**"
- Scrub Right Keyboard shortcut: "**Alt + Shift + Arrow right**"

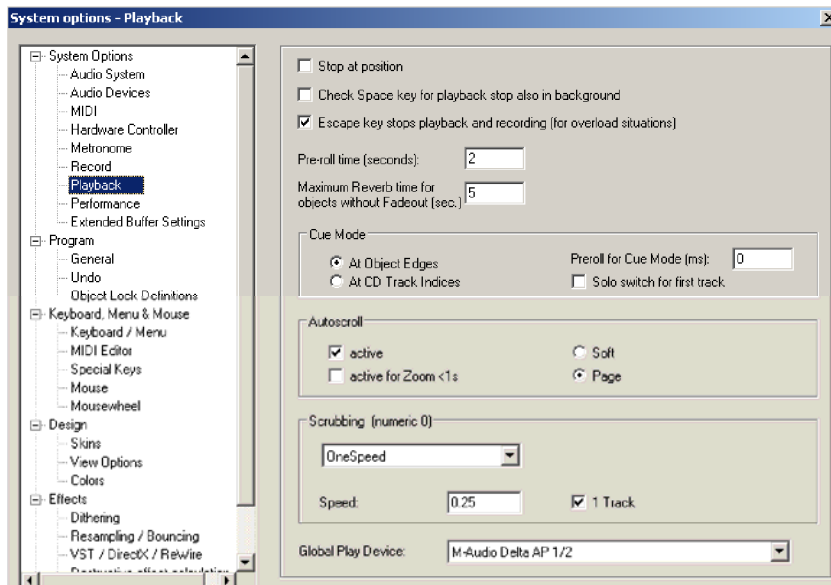
Detailed information about scrubbing is provided in "Playback -> Playback options -> Scrubbing (view page 579)"

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.

Playback parameters

Choose this option to open the play parameter window.



Shortcut: P

Check the **"Stop at current position"** option if you wish the play cursor to remain at the same position after stopping playback. If this option is not active, the play cursor will jump to the original position or back to the start of the range when playback is stopped.

The **"Check space key for playback stop also in background"** option also allows the space bar to be used for "Play" and "Stop" in Samplitude while using a different software application.

An additional function can be used in case of overload, i.e. **"Escape key stops playback and record"**.

The **pre-roll time** is a necessary parameter for editing in Samplitude. Pre-roll time is used to define (to the second) the desired section before the start of the selected range.

The **maximum reverb time** for objects without fade-out can be set up to 60 seconds here.

Note: Please note that the length of reverb time can lead to performance problems.

Cue ensures that the **first track in cue mode is switched to "Solo"**. **Pre-roll (ms)** may be used to specify in milliseconds how long the delay between execution of the record command and the actual start of the recording should be.

Detailed information about cue mode is available via the menu item "Play / Rec -> Playback mode -> Cue mode (view page 576)".

With **Autoscroll** the graphical display runs the entire time before the play cursor reaches the visible section. This provides a constant overview. Switch to autoscroll mode by checking "**Active**". "**Active for zoom < 1 s**" causes autoscrolling to begin at very high zoom levels of less a second

Choose between "**Page**" and "**Soft**" autoscroll modes. With page scrolling, the section changes before the play cursor moves outside the section, while with soft scrolling, the play cursor always remains in the middles of your chosen section and the arrangement moves along beneath it. At smaller buffer sizes (view page 25) (e.g. < 4096 samples), scrolling will be softer.

Note: The autoscroll process can overload the processor in some cases, in which case dropouts will occur in 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 play cursor and the mouse position is used to control the speed. The play cursor follows the movement of the mouse. The further you move the mouse from the play cursor, 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.

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 80)".

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 at the play cursor position.

If you have selected a range behind the play cursor on the grid and marker bar and have activated "Loop" mode, then recording will only begin at the loop start. Click the recording button at the play cursor position and the track will signal that it is ready to record (the recording button will blink during playback). Recording will only be activated once the start of the range is reached, and the selected range will be recorded in "Loop" mode.



Record without Playback

If you begin recording in "**Record without playback**" mode, then recording will start at the play cursor position when the record button is pressed. Initially, only the track's input signal will be audible at the point where you are currently working in "Record" mode. The play cursor will not run in the timeline.

If playback is now started from the position of the play cursor, all playback tracks will be audible from this position. Recording continues independently until the "Record" button is pressed again. The recording that was just created will now be visible as a new object in the arranger.

Punch Marker Mode

This button activates Punch Marker Mode. "Punch-in/out" is a recording process which 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 may be started (punch-in) and stopped again (punch-out) at any time during playback. Multiple punch processes may be made during a single pass in order to improve various positions in one recording. Start playback with the spacebar. Next, clicking the "Record" button in the transport console "punches in" and "punches out" again.

Note: If you check "Prepare all tracks for track punch record" in the "System options -> Record (view page 475)" menu, you will also be able to punch your recording by pressing the "R" 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 the "R" key of the corresponding track. The actual recording occurs within the punch range. While this is happening, the "Record" button will flash when the play cursor is in front of the "In" marker. The button remains red during the punch recording.
- **Multiple punch recordings in one pass:** It's possible to make multiple punch recordings in one pass with the help of punch markers. To do so, add extra punch markers by positioning the play cursor at the desired point and, while keeping the "Alt" key held, press the "In/Out" buttons in the transport control.

Note: When working with punch processes, it is useful to have "Auto crossfade" mode active. "Auto crossfade" will create smooth transitions automatically between takes so that no pops or crackles result.

- **Punch recordings** may also be executed **as loops**. Select a range across the planned punch range, 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 150)".

- If the recording option "**Pre-activate all tracks for track-punch recording**" is active ("System options -> Recording"), then audio tracks that were not active when recording was started may be added to a recording, or tracks may be removed from the recording. Click the "Record" button for the desired track. Punching an individual track requires that the track has been assigned to a sound card input that has not already been used for recording. This is indicated by a red circle around the record button in the track header.

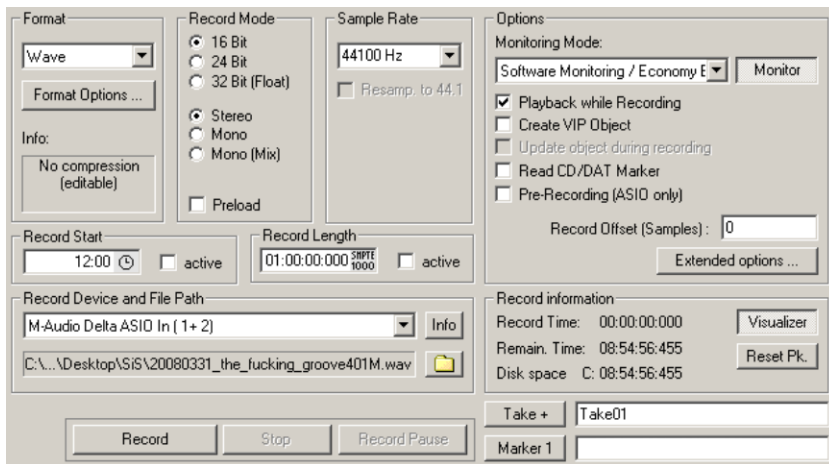
Note: Tracks that have been displaced in "Punch marker" mode via individual track punching will not be influenced by punch markers.

Record Pause

Recording is set to pause.

Record 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 483)".

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.

Further recording Options

Monitoring Mode:

Please also read the chapter "System options -> Global audio options -> Monitoring settings (view page 27)" for more details.

Monitor: Use this button to activate the LED control displays. Please note that the recording device must be selected prior to beginning this. While recording, the LED control displays move slower, but they nevertheless show each maximum level.

Play while recording: Activate synchronous recording and playing here if this supported by your sound card.

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 whether or not all recordings should be integrated automatically into a virtual project via this mode. 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 inserts audio material added at the beginning of the recording to the beginning of the current recording.

More information about pre-recording is available via the menu item "Program preferences > System/Audio > Record (view page 475)".

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 via File > Program preferences > System/Audio > Record (view page 475)".

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: During active external synchronization as the "Slave", recording does not start immediately, but rather when the master starts.

Stop: Ends a running recording process.

Recording pause: Use this button to interrupt the recording process. The play cursor continues to run. Pressing "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 150) 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 play cursor, e. g. to correct an erroneous passage 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 27)" for more details.

Move Play Cursor to Beginning

This option positions the play cursor at the beginning of the project.

Shortcut: Home

to End

This option positions the play cursor at the end of the project.

Shortcut: End

to Range Start

Sets the start position of the play cursor at the beginning of the currently selected range.

to Range End

Positions the start position of the play cursor at the end of the currently selected range.

to Start of Section

This option positions the play cursor at the beginning of the visible section.

Left/Right Move in Page/Scroll Mode

Use these commands to move the play cursor to the right or left.

In "Page" mode, the play cursor 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 play cursor jumps to the middle of the section and remains there as the project moves through it.

Shortcuts:

Movement in "Page" mode:

Right arrow/left arrow

Movement in "Scroll" mode:

Alt + left/right arrow

Note: Once a range open is open, use the "left/right arrow" keys to control the beginning of the range.

Object Border Left

The play cursor jumps to the left to the next object edge in the selected track.

Shortcut: Ctrl + Q

Object Border Right

The play cursor jumps forward to the right to the next object edge in the selected track.

Shortcut: Ctrl + W

Marker Left

The play cursor jumps to the left to the next marker in the arranger window.

Shortcut: Alt + Q

Marker Right

The play cursor 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 play cursor may be positioned at the last stop position with this command nevertheless.

Shortcut: Ctrl + Alt + ,

Repeat Last Position(s)

This command jumps the play cursor to a maximum of 5 previous stop positions.

Shortcut: Backspace

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.

Save marker

With the option "**Marker with name...**" you can specify further markers to which you can assign a name of your choice.

In the "**Marker with names and numbering...**" dialog " you can freely number and name additional markers. The "-1 sec" button drags the marker one second forward in the marker list.

The command "**Markers with automatic numbering**" automatically creates markers with sequential numbers.

Shortcut: Shift + #

The function "**Marker on record position**" causes a marker to be created 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 play cursor's current position by selecting a number between one and ten from the menu.

Shortcut: Shift + 1...0

Jump to markers 1 - 10: You can set **marker positions** at the play cursor's current position by selecting a number between one and ten from the menu.

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

Deletes all markers of the selected range.

Set New Audio Marker

This command sets an audio marker at the current play cursor 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.

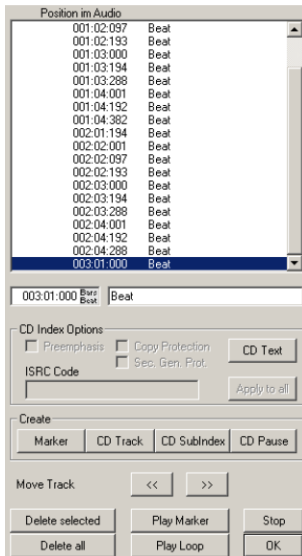
Make Audio markers visible by marking the "Audio markers" checkbox in the view options ("Shift + Tab") in the "Objects" section.

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.

Set new Tempo marker

Detailed information on placing Tempo markers can be found in the chapter Edit > Tempo > Tempo Map Dialog". (view page 301)

Set New Time Signature marker

Detailed information on placing time signature markers can be found in the chapter Edit > Tempo > Tempo Map Dialog". (view page 301)

Set new grid marker positions

Detailed information on grid position markers can be found in the chapter Edit > Tempo > Tempo Map Dialog". (view page 301)

Ignore all Tempo Markers, Use Project Tempo

This option enables all previously created tempo markers in your project to be ignored so that only the project tempo is authoritative.

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".

Auto JamSession

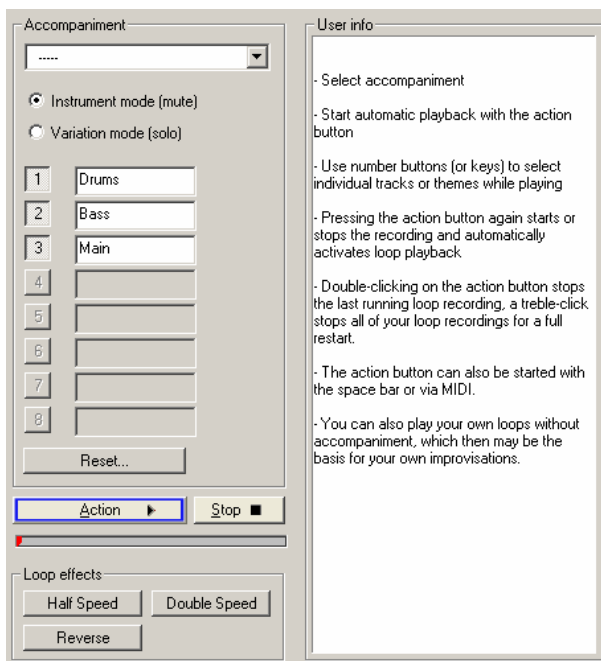
You can now jam quickly and creatively with your virtual band thanks to Auto Jam Session's practical loop automation function. Record your instrument on separate tracks and develop the song structure systematically.

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 play cursor position or
- stop all imported loops at the play cursor position or
- stop all loops at the play cursor 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 play cursor 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.

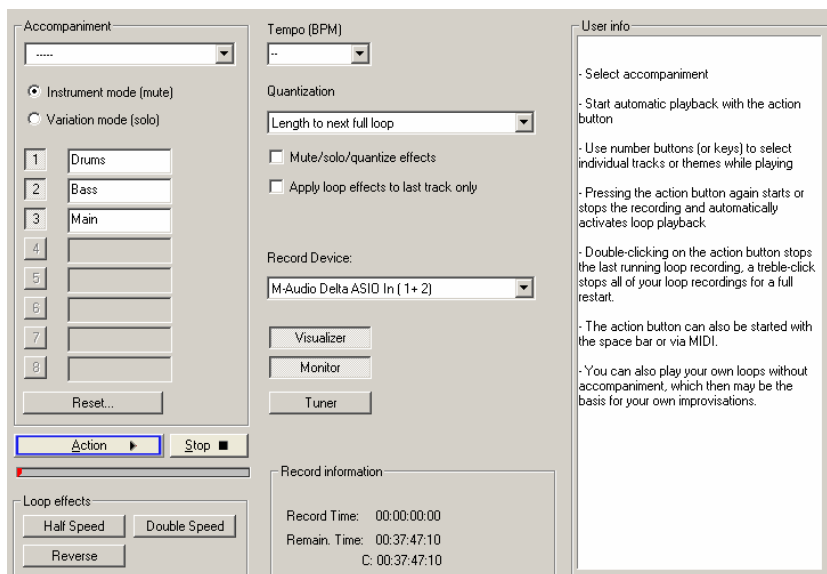
Double speed: Press this button to double the playback speed.

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).

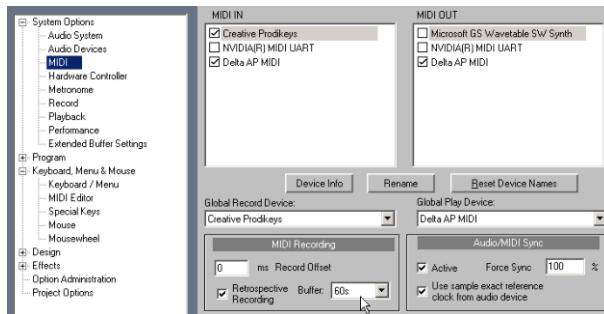
Auto JamSession - recording information

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

No effect (Track)

If you select this display option the track automations are deactivated.

Edit selected curve

Create

The curve generator (view page 335) creates a beat-based volume curve and pulses according to eighth, quarter, half, or full notes. You can enter a minimum and maximum value or define delay values as an option for the beat-based envelope curve calculation.

As an alternative, you can also let the curve generator write an envelope curve scan.

Invert

This command inverts the activated automation curves.

Thin Out

This function "thins out" the activated automation curve by reducing the number of automation events. During recording, the automation events are placed in very short intervals. The "Thin out" command reduces the number of curve points. The course of the automation is then displayed and reproduced more accurately.

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.

Copy, Paste, Delete

The active automation curve may also be copied, pasted into other tracks, and deleted.

Delete All Curves

This command deletes all automation curves in the corresponding track.

(Track) Automation Mode

Specify the track automation mode here. "**Copy to All Tracks**" applies the currently selected automation mode to all tracks/channels. Select "**Write Only Existing Curves**" to apply the automation data to existing automation curves only.

Detailed information about the automation modes is available in the chapter "Automation -> Automation modes (view page 329)".

MIDI Controller Automation

This opens the "MIDI controller/automation curve settings" dialog.

Detailed information about MIDI controllers is available in the chapter "MIDI controller automation (view page 338)".

Shortcut: Ctrl + Alt + A

Hide Automation

Use this option to deactivate the display of automation curves for all tracks.

Show Track Automation

Select this display option to display only the track automation.

Show Object Automation

If you select this display option, only the object automation will be displayed for all tracks.

Show Only Selected Curve

Views only the selected curves for the corresponding track. This helps to maintain a clear overview if several automation curves have been created.

Show all curves (not selectable)

The deselected automation curves are grayed-out. They cannot be activated with the mouse.

Show all curves (selectable)

The deselected curves are also shaded, but may now be activated with the mouse.

Select previous curve

With this you can select the last edited curve.

Select next curve

With this you can call up the next curve from the selection list.

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.

Notes concerning offline effects editing

- When editing objects via the context menu item "**Edit wave project**", 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 Wave Content**" 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 in the VIP later (this allows you to undo destructive editing).
- The following functions are available exclusively for wave projects: "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 497)" for virtual projects and for wave projects in order to activate the undo function.

- Please note that most functions only work within the selected range in the wave window. To use the functions on the entire wave project, the entire wave project must be selected (Shortcut: "A"; menu item: "Edit > 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. (Eine Ausnahme bildet hier nur die Funktion „Noise Sample holen"). 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.

Play/Stop: This button starts/ends the preview function for each offline effect. Previewing editing does not occur in real-time, the result, however, will match the end result.

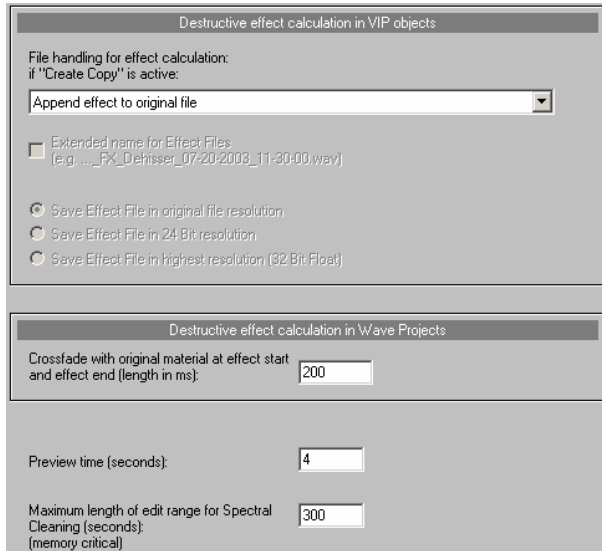
OK: The edit is applied to the selected range of the sample or the selected object.

Cancel: Click this button to close the dialog window without applying any editing.

Help: Displays the Help feature.

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 "Options -> 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, normalize (quick access), loudness adaptation

Dynamik: Dynamics, Advanced Dynamics, Multiband Dynamics, sMax11, AM-Munition (Samplitude Pro X Suite), AM-Track (Samplitude Pro X Suite / SE Version for Samplitude Pro X), AM-Phibia (Samplitude Pro X Suite), AM-Pulse (Samplitude Pro X Suite)

Frequency/Filter: Parametric EQ, EQ116, FFT filter/spectral analysis, brilliance enhancer (optional), Filtox

Delay/Reverb: Delay, room simulator, Ecox, Variverb

Time/Pitch: Resampling/Timestretching, Elastic Audio

Distortion: Distortion, Amp simulation, Vandal (Samplitude Pro X Suite / SE Version for Samplitude Pro X)

Restoration: De-clipper, de-clicker/de-crackler (optional), de-hisser, de-noiser, get noise sample, remove DC offset (offline), spectral cleaning (offline)

Stereo/phase: Switch channels, multiband stereo enhancer, invert phase (both channels, left channel, right channel)

Modulation/Special: Convolution, vocoder, Corvex, reverse

Sample manipulation: Change sample rate (offline), sample number/2 (offline), sample number *2 (offline), reverse, form loop (offline)

MAGIX Plug-ins: AM-Munition (Samplitude Pro X Suite) AM-Track (Samplitude Pro X Suite/SE Version for Samplitude Pro X), AM-Phibia (Samplitude Pro X Suite), AM-Pulse (Samplitude Pro X Suite), Corvex, Ecox, Filttox, VariVerb, Vandal (Samplitude Pro X Suite/SE Version for Samplitude Pro X)

essentialFX: eFX_ChorusFlanger, eFX_Phaser, eFX_Reverb, eFX_StereoDelay, eFX_Compressor, eFX_Gate, eFX_DeEsser, eFX_VocalStrip, eFX_TubeStage, eFX_TremoloPan.

Process Only Left Stereo Channel

Process Only Right Stereo Channel

Process Effects Offline: Tick the checkbox here to process effects offline

At object level you can swap object channels, invert phases, Object EQ, Object Dynamics, Pitchshifting/Timestretching, Object backwards, Elastic Audio, and MAGIX plug-ins.

Amplitude

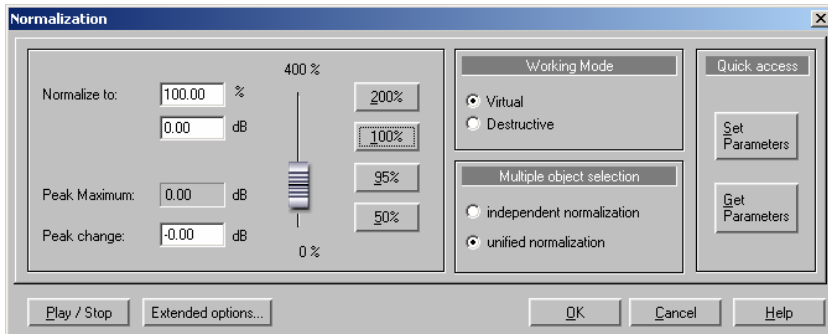
Normalize

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 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 are 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.

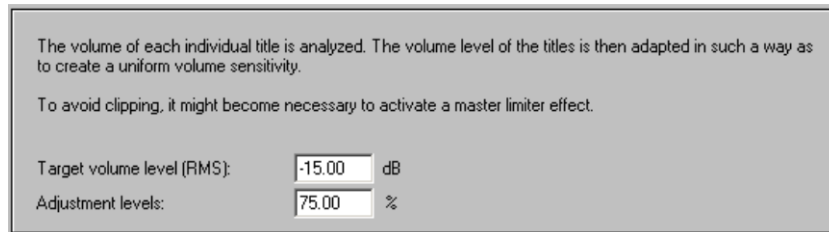
Normalize (Quick Access)

This function enables the specified settings in the "Normalize" dialog under "Quick access - > Set parameters" to be used immediately.

Shortcut: N

Loudness Adaption

This command executes an automatic RMS normalization. The loudness of each individual title is analyzed and the level of the titles is adapted to create uniform volume sensitivity. The preset target value is -15 dBfs. The target value may also be adjusted individually. The closer the value is to 0 dBfs, the louder the signal will be.



This adjustment determines the percentage of control between the titles and the target volume 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. **essentialFX Compressor** (view page 690): A simple, efficient tool for reducing dynamics with soft characteristics and adaptive control processing. The "essentialFX Compressor" is a very musical compressor.
2. **Dynamics** (view page 607): A quick and easy-to-use dynamics module. Its main functions (setup) may be set to compressor, expander, gate, or limiter. More specific parameters may also be changed; "dynamics" always uses an efficient algorithm, which also reduces required CPU resources.
3. **Advanced Dynamics** (view page 608): This module is a very comprehensive tool. It's 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 via direct input or by graphically changing the characteristic. Level detection may be derived via either "**Peak**", "**RMS**", or "**Fast**" mode.
4. **Multiband Dynamics** (view page 612): 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, the function is able to prevent bass peaks from reducing the entire signal. Multiband technology also lets you specifically edit individual frequency ranges.
5. 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 698)".
6. **sMax11** (view page 618): This module enables the loudness of the audio signal to be increased by specifying input amplification (gain in). Essentially, this involves a hard or brickwall limiter with input amplification.

Which compressor should be used and when?

When to use a certain dynamics module depends on how specific the change needs to be. The "Dynamics" and the "essential FX Compressor" modules work well on mixer channels to quickly and efficiently influence the dynamics.

The sound of an object should normally be changed very specifically. In this case, the dynamics of an instrument may be varied between the verse and the chorus, depending on the playback method. This is a desired relation that may 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.

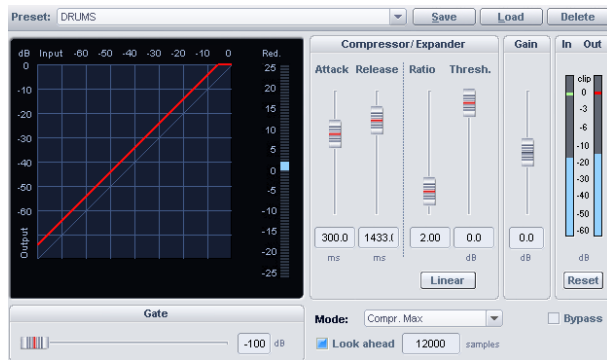
Experiment with "AM-Track", "AM-Pulse", "AM-Phibia", and "AM-Munition" to add characteristic pre-amps or tube sounds to your pieces.

The mixer's master module features "Multiband dynamics" to give the entire mix its final balance. Since increasing the volume using a multiband compressor may produce clipping, it's necessary to have a limiter, which then approximates all clipping samples at the limiter threshold.

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.

"sMax" is intended as a loudness maximizer/limited that adds increased loudness to your audio signal.

Dynamics (Compressor/Expander/Gate/Limiter)



Use this editor to adjust a sample's dynamics. Each function may be previewed in real time.

The graphics display shows the corresponding resulting dynamics curve for an improved overview.

The following functions are available:

Compressor: The dynamics of a piece are limited, loud sequences are dampened, quiet sequences become louder. Compression is often used to add power to audio material or to increase the loudness. The level of compression is adjusted by the "Ratio" controller; the activation threshold is determined by "Threshold". You can also influence the "Attack" and "Release".

Limiter: Only the loudest passages are limited (above the threshold). Quiet passages remain unchanged. Limiters are used to reduce the level peaks without reducing the master dynamics. After the limiter process, the master level can be increased using normalization without having to worry about overmodulation. The "Limiter" preset sets a threshold of -0.2 dB when using the limiter as a peak limiter in the master area of the mixer.

Limiters 100%: Performs the same editing as the limiter; however, the level is immediately raised to 0 dB, which corresponds with a subsequent normalization.

Expander: The dynamics of a piece are increased, loud sequences remain quiet, quiet sequences become even quieter. Dynamics expansion is often used for recording speech featuring 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 De-noising algorithm provided.

Gate: 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.

Dynamics - Parameters

Ratio: This parameter controls the compression ratio. A value of 5.0, for example, corresponds to a volume reduction that's five times the threshold, while a value of 1.0 means that compression is not applied.

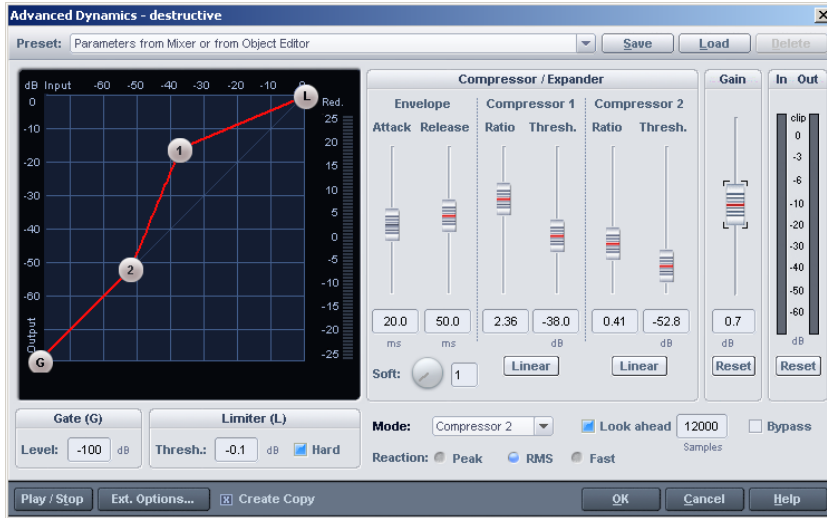
Threshold: This sets the threshold; the effect will be applied above and below this level.

Attack: This sets the time frame from crossing the threshold to the maximum extent of the effect.

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

Gate level: This parameter specifies which levels should be set to 0.

Advanced Dynamics



Advanced dynamics – general controls and presets

Load, save, delete presets: Save, load, or delete settings here. The default file ends with *.dy2. In addition, all dynamics presets (*.dyn) can be loaded. The parameters are converted to those available in the "Advanced dynamics" module. If saved again, the preset will be saved with the *.dy2 extension to make sure the settings are stored.

Dynamic scope: When the selected object/sample is played, the in and out level of the signal appears as a white, scattered line.

Reduction display: The level reduction of the played audio material is displayed to the right of the graphic display.

In/out displays: These meters show the input and output displays in dB.

Reset: Reset the reduction, in and out displays.

Mode: These setups are typical applications of "Advanced dynamics". They provide default settings that make the graphical editing of the reference line easier. In this case, the number of editable parameters are limited by the choice of the setup so that with pure limiter application, for example, the other curve points will not change.

OK: Accepts the parameters.

Cancel: Rejects the parameters.

Advanced dynamics – editing the reference line (graphically)

To graphically edit the static transmission reference line, left click one of the bases of the curve on the reference line and move it while holding down the mouse button. Moving the bases at a point often also influences the parameters of other points.

To position the point directly in the reference line field, right click. A dialog appears to read and enter the input and output values.

If you switch on the "Hard" parameter in the limiter, a blue field entitled "Limited zone" will appear for threshold values below 0 dB. This field shows that the level of the signal never extends into this limited zone.

In order to hide a handle, move the applicable threshold slider to 0 dB. In order to make it visible again, the threshold must be below the limiter threshold.

Advanced dynamics – editing the reference line (parametrically)

Gate

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

Level [in dB] determines the maximum level at the output. Note the "Hard" setting.

Hard: When this parameter is active, the output signal limits exactly to the set threshold value. This means that no value shall be higher than the set limiter level. Of course, the signal is not simply cut off at this limit; the applied algorithm brings the signal as close as possible to the limit without changing the sound much. 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, in which case 0 indicates 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: This sets the time frame between the crossing of the threshold and the maximum extent of the effect in milliseconds.

Release: This sets the time frame between reduction of the threshold and the complete withdrawal of the effect in milliseconds.

Compressor 1/2 threshold: This sets the input threshold in dB; above this level, the respective component will be activated.

Compressor 1/2 ratio: This parameter controls the compression ratio. 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. Please note that the parameters influence one another.

Advanced Dynamics - Dynamic parameter

The "**Reaction**" function works much in the same way as the Compressor does to the source material.

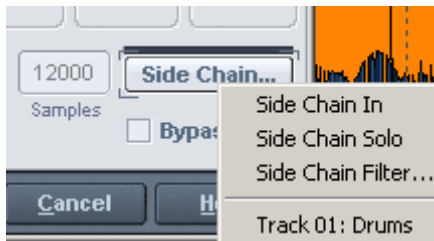
PEAK: This mode uses the signal's peaks as a basis for calculation. The dynamics module reacts to level peaks quickly and efficiently.

RMS (Root Mean Square): This mode 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 only be modified a little. The maximum level at the output is never higher than the limiter.

Preview: If this option is not checked, the dynamics section will not provide a preview. Thereby allowing artifacts (pumping) and over modulations to be set. 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 (view page 621).

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" with individual points that represent the input and output values. This way, the change to the samples achieved via advanced dynamics may be visualized constantly.

The dynamic scope's functions may be easily explained via the following example:

Select "ZERO" and press "Play", now the effect won't change the signal. The scope shows all samples on the red line, which indicates that all output values equal the input values. Slide the gain fader upwards to see that the "snow cloud" follows the reference line. If very high time constants are selected, then 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.

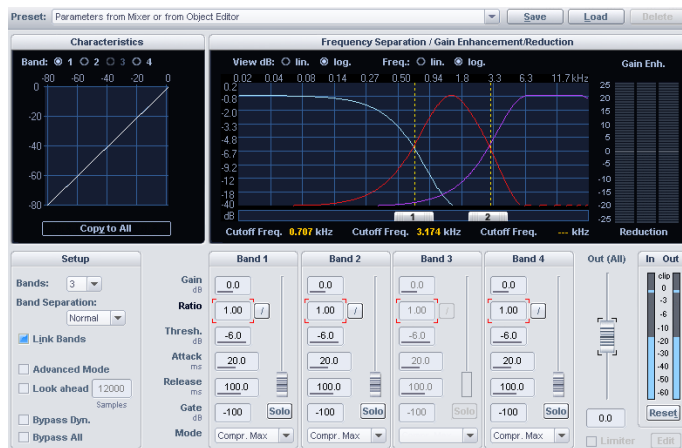
Advanced dynamics – tips and tricks

- 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 shows you the frequency response curves of the individual curves. The left axis labeling shows the damping in -dB, while the upper axis labeling shows the frequencies in kHz. The individual curves are displayed in color.

The illustration to the right displays the dynamics reference line. It illustrates the ratio from the input level (upper axis in dB) to the output level (left axis in dB).

The dynamics reference line is always displayed for the band that was selected for adjusting the dynamics.

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

Saving, loading & delete presets: This is where all parameters are saved loaded or deleted. The default file extension is *.mdy.

Reference line

Band: Select a band for dynamic editing here. The reference line of each selected band is displayed in white, while the other bands are displayed as lines in their corresponding colors.

Copy to all bands: After pressing this button the parameters of the selected band, i.e. the parameter values that 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: Select the number of bands between 1 and 4. On setting 1, there is no separation into various frequency bands. In this case the algorithm works like the standard dynamic section. The more bands there are, the more processing power the algorithm requires.

Band separation: See below

Link bands: If this option is active, the changes to the dynamic parameters affect all bands. It's usually unnecessary to set up the dynamic parameters for each band individually. Especially with general settings, it's often necessary to adjust the parameters together.

Advanced mode: If this option is selected, the same internal processing routines are set as in the "Advanced dynamics" module.

Note: If you have activated "Link bands", this 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: If this option is checked, the dynamics section will perform a preview. When unchecked, artifacts like "pumping" or distortion may occur. 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):

The dynamic editing of individual bands is removed from the signal route. This option is used to compare the results of editing with the unchanged original.

The difference from the "Bypass all" switch is that when "Bypass dyn." is in solo mode, individual bands may be compared separately, with or without dynamic editing as the filter bank is still activated.

Bypass all:

In this case as well, the entire multiband dynamic section is removed from the signal route. Since the entire filter bank is "bridged", "Solo" mode may no longer be used. The gain fader for the sum is also no longer active.

Help: Displays the "Help" feature.

OK: The algorithm is applied to the selected range of the sample or the selected object. The parameters of the dialog window are applied when used in the master section of the mixer.

Cancel: Click on this button to close the dialog window without calculating the effect. When used in the master section of the mixer, the parameters of the dialog window are not applied.

Multiband dynamics – setups and parameters

Parameters for the dynamic sections

Ratio (factor): This parameter controls the level 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): This sets the time frame from crossing the threshold to the maximum extent of the effect.

Release (ms): This sets the time frame between the point below the threshold and the complete reduction of the effect.

Gate level (-dB): This parameter specifies the value after which the level should be set to 0.

Output parameters:

Gain Band (dB): This adjusts the level at the output of the dynamics stage individually for each band. Note that the influence of this controller cannot be fully displayed. If you set the controller higher, clipping may occur; in this case, the reference line continues upwards out of the display border.

Solo mode: In "Solo" mode, frequency bands may be monitored one at a time. This function particularly simplifies the setting of filter parameters. For instance, you can locate a critical frequency range of a mix before editing the dynamics.

Mode

Compr. max: The dynamics of a frequency range are limited, loud sequences are dampened, quiet sequences are retained. These settings are often used to add power 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 influence of the dynamics corresponds with the behavior of a "classic" compressor: The dynamics of a frequency range are limited, loud sequences are dampened, quiet sequences are retained. This setting is used especially for sound design via compression without raising the total volume. The level of compression is adjusted by the "Ratio" controller; the activation threshold is determined by "Threshold". "Attack" and "Release" may also be influenced. A "Gain" controller has also been added to the controller to compensate for volume reduction.

Expander: The dynamics of a frequency band are increased, loud sequences remain quiet, quiet sequences become even quieter. Dynamics expansion is often used for recording speech featuring a high noise level. The expansion causes the level of the speech to be raised while the noise is suppressed. Note that a powerful De-noising algorithm is also provided for noise suppression.

Gate: This function dampens or sets very quiet passages below the threshold level 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 level peaks without reducing the master dynamics.

Limiter 100%: Performs the same editing as the limiter; however, the level is immediately raised to 0 dB, which corresponds with a subsequent normalization.

Note: To use the limiter as protection against overmodulation, you should take into account that the limiter may only reliably prevent overmodulation in a single band. If the bands are mixed together, the sum may again result in an excess of the threshold level set for the bands.

Out (all)

Set the volume level for the entire algorithm here. This parameter cannot be set individually for each band. The graphic does not reflect this setting. Use the "Volume" fader to balance the volume difference via the dynamics process.

Limiter on/edit (only in the master area of the mixer and in the object editor):

This turns off/edits the peak limiter. It prevents overmodulation of the volume increases when using the multiband dynamics section.

Separation frequencies

Use the sliders at the bottom of the graphic to adjust the separation frequencies for each of the filter bands. The number of separation frequencies depends on the number of selected bands ("Bands" parameter). The border frequencies for the deepest and the highest bands (high-cut and low-cut) are displayed. The border frequencies are those at which filter damping amounts to -3 dB. For mid bands (band pass 1 and band pass 2), the mid frequencies and bandwidths are displayed. The bandwidth is, in this case, the distance between both separation frequencies. This corresponds to the intersection 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 (setting low: approx. 25 - 35 dB, normal: approx. 35 - 45 dB, high approx. 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. Click "Copy to all bands". Selecting this mode ensures that all bands feature the applied setting.
2. Activate "Link bands". Any changes you perform in the current band are automatically mirrored on the other bands.
3. 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".

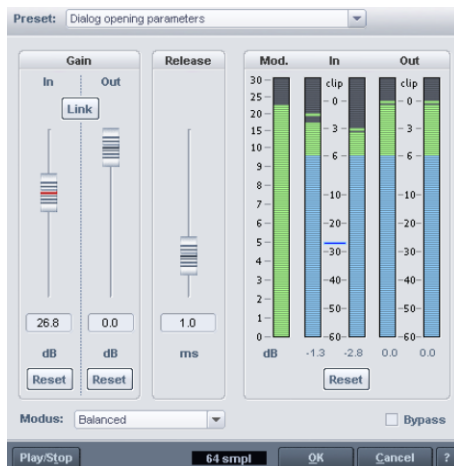
A third option is to first compare the supplied presets and then use the presets as a starting point for searching for optimum settings.

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 increases 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 approach might look like this: First, switch on the **link** option. Now increase the input amplitude until the audio changes become unacceptable. Reduce the level again slightly. Distortions may be minimized by increasing the release time. In any case, the compression effect and the increase in loudness will therefore be lower. Maybe switch on the bypass. The link option keeps the loudness constant, allowing changes to be compared better. Once the optimal setting has been located, switch off the link option, and set the output gain back to 0 dB.

sMax11 - general controls and presets

The options **Save**, **Load**, and **Delete** are not provided directly in the dialog for this effect. These are integrated into the preset list, at the bottom. Click these list entries to load, save, or delete presets. The default file extension is ***.max**.

Play/stop: This button starts/stops the real-time preview function. Use it to check the corresponding filter setting.

Latency display: This value provides information about current distortion of the effect in samples that should be included for latency compensation. The latency of this effect is influenced by the set effects mode (among other things).

OK: Applies the preset parameters.

Cancel: Rejects the parameter 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 control elements

Gain-in: This value specifies how much the signal should be increased (maximum).

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, e.g. without increasing the output volume. This makes it easier to detect distortions resulting from a high "Gain-in" value.

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:

- **Balanced:** Lowest level of distortion with transparent sound, robust to use.
- **Fast:** Accesses slightly faster than "Balanced". This mode corresponds with the behavior of the hard limiter in the advanced dynamics unit.
- **Aggressive:** Very short attack time, suitable for especially sharp transients for this reason
- **Hard clipper:** Controls are shut off in this mode. Level peaks are simply cut off, which may lead to serious distortion. This mode may be useful if transients of an individual hit still need to be highlighted.

Use of different modes is dependent on the case of application. For example, "Balanced" mode should be used for speech. For signals with dominant percussive sections, the sharpness of the hits may be maintained by using "Aggressive". The right setting will cover distortions that occur or make them very minimally audible.

efx_Compressor

The **(essential) eFX Compressor (view page 690)** is a simple, efficient tool for reducing dynamics with soft characteristics and adaptive control processing. It is a very musical compressor.

AM-Munition, AM-Track, AM-Phibia, AM-Pulse

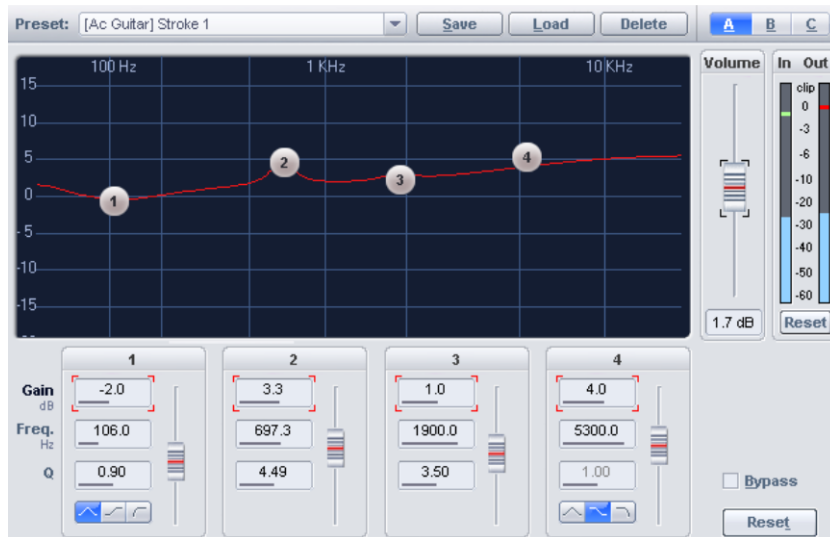
(Samplitude MX Suite)

More information is available in "MAGIX plug-ins (view page 698)".

Frequency/Filters

Parametric EQ

This dialog features a parametric 4-band equalizer.



The parametric equalizer allows you to activate filters for four freely selectable frequency bands to adapt the sound of a sample/object more individually. Broadband elevations in high and bass values as well as narrowband corrections to the frequency response can be carried out here.

Volume: Use this controller to adjust the master volume by up to +/-20 dB should the level rise or sink too much while filtering.

Gain: These controllers allow you to raise or lower the filter +/- 20dB. 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).

Note: Depending on the parameter you select (gain, frequency, or Q factor), the fader to the left of the input fields controls the corresponding parameters. Gain and frequency values can also be changed directly in the display by dragging with the mouse.

Type: Set the filter type here. Filters types available include **Peak**, **Shelving**, **High pass (= low cut)**, and **Low pass (= high cut)**.

Note:

The shelving filter starts signal manipulation at the cutoff frequency in order to apply the effect until the end of the audible range. This includes high and low-pass filters.

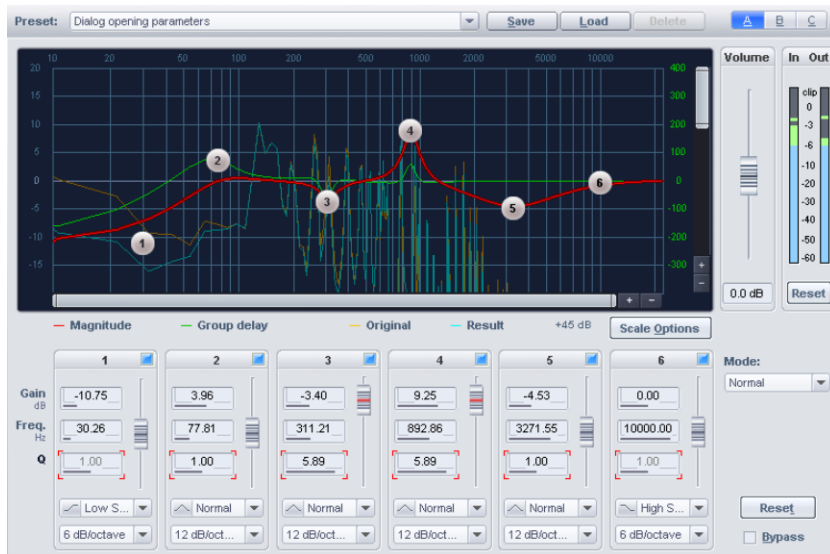
A high-pass filter cuts the signal sharply after the defined cutoff frequency.

A low-pass filter cuts the signal sharply above the defined cutoff frequency.

The peak filter stretches the around the defined cutoff frequency. This includes "normal" parametric EQ, bandpass filters, and notch filters.

Play/Stop: This button starts/stops the realtime preview function. This lets you check the acoustics of the respective filter settings.

Setup A, B, C: Switch between three different filter setups using the mouse to quickly execute an acoustic comparison of two different settings, even during previewing. Save the individual EQ setups with a mouse click on the corresponding button (A, B, or C) while holding down "Shift".

EQ116

The EQ116 provides filters for six freely selectable frequency bands to adapt the sound of a sample/object more individually. Broadband elevations in high and bass values as well as narrowband corrections to the frequency response may be applied here.

EQ116 - general control elements, presets and setups

Load, save, delete presets: Save, load, or delete settings here. The default file extension is *.aeq.

Setup A, B, C: Switch between three different filter setups using the mouse to quickly compare two different settings, even during previewing.

Reset: Sets the effect to its original initialization state.

Bypass: The algorithm is removed from the signal route. This allows the unedited signal to be compared to the result of the algorithm.

Play/stop: This button starts/stops the real-time preview function. Use it to check the corresponding filter setting.

Latency display: This value provides information about current distortion of the effect in samples that should be included for latency compensation. The latency of this effect is influenced by the set effects mode (among other things).

OK: Accepts the parameters.

Cancel: Rejects the parameters.

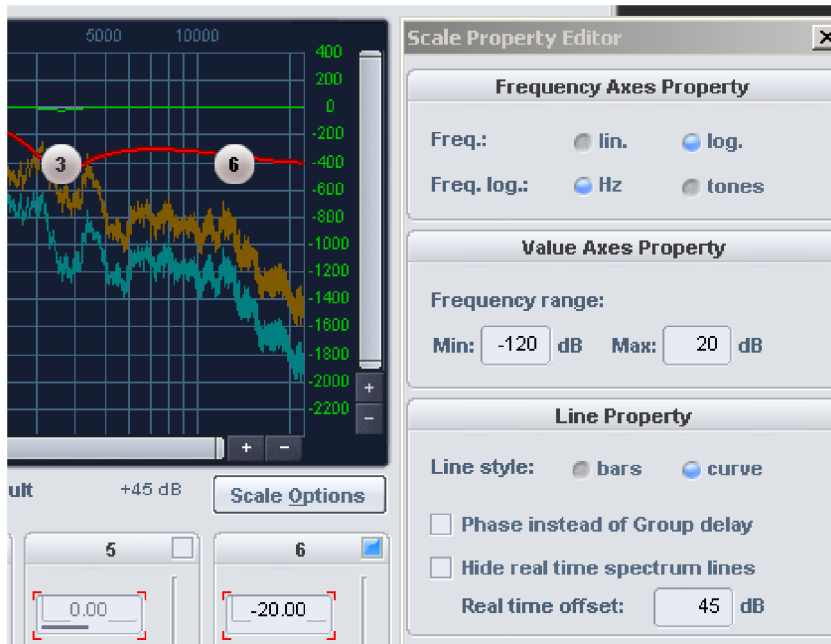
Help: Opens the "Help" file for this effects dialog.

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 level points. The bandwidth can be changed by turning the mouse wheel near to a level point

Volume: Use this controller to adjust the master volume if the level rises or sinks too much while filtering.

Tape on/off: In the header for each tape group, there is an on/off switch for turning the corresponding tape on or off. The associated level point will disappear from the filter graphic accordingly. In der Filtergrafik verschwindet dann auch der zugehörige Stützpunkt

Gain: These controllers allow you to raise or lower the filter. Setting the controller to 0 deactivates the filter and doesn't use 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 a **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**, there is a choice available between 6 dB/octave and 12 dB/octave. The filter type "**Normal**" is fixed at 12 dB/octave.

Note: Depending on the parameter you select (gain, frequency, or Q factor), the fader to the left of the input fields controls the corresponding parameters.

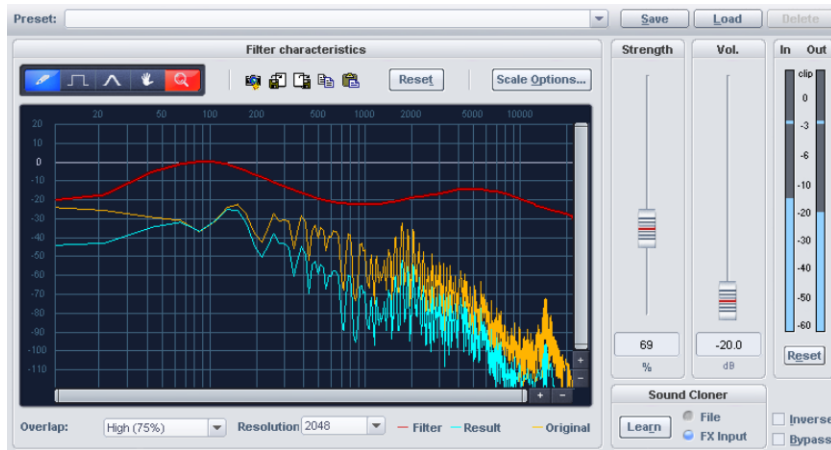
EQ116 - internal operating modes

Mode: Set the internal operating mode for the EQ116 here.

The following modes are supported:

- **Normal:** The advanced equalizer operates in a manner that corresponds with the parametric 4-band equalizer, and therefore represents a compatibility mode.
- **Oversampling:** This effect works internally compared to the first mode and a higher rate at which the filter formation takes place. In particular, the upper frequency range achieves a better amplitude frequency response.
- **Linear phase:** This filter formation occurs such that no phase shifts occur. Processing of this filter takes place according to a process that is fundamentally different than other modes.

FFT Filter/Sound Cloner



The FFT filter (for real time use in the object editor, in the track or master section) allows you to precisely filter linear signals. The FFT filter for real-time application is a "minor" option with a huge number of possible settings, analyses, and functional modes for destructive editing of audio material.

Note:All effects opened using this menu are calculated destructively, provided that, additionally, the option "**Process Effects Offline**" is active. You do, however, have the option to work with a copy to be able to keep the original audio material. The "**Create copy**" option is already selected in the corresponding dialog.

Edit the red filter curves by drawing with the pencil tools in the graphical display. If you let the object play along, you will immediately hear how the sound of the edited object.

The graphical view shows 3 curves that are integrated in a coordinates system. This includes the vertical axis, which shows the volume in dB, and the horizontal axis, which shows the frequency in Hertz (the pitch).

The **yellow curve** represents the **original frequency response**, and the **blue one** represents the **corrected frequency band** that accompanies the original frequency response after the **red filter curve** is applied.

The following tools are available via the object FFT filter:



Pen for free-hand drawing: This tool allows you to draw any curve you need. Hold the "Shift" key down to draw straight lines.

Pen for quantization 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 may be enhanced by clicking by holding down "Ctrl".

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 in 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 (Shortcut: Ctrl + C).

Insert: Use this command to insert the filter curve from the clipboard into the FFT filter selected for the 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

Freq.: lin. log.

Freq. log.: Hz tones

Value Axes Property

Zoom ratio of edit-line:
 none 5:1 10:1 20:1

Scaling dB: lin. log.

Frequency range:
 Min: dB Max: dB

Line Property

Line style: bars curve

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.

Value display: 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 may be defined freely. Simply enter the desired minimum and maximum values into the corresponding fields.

Curve display: Display the curve roughly as a stepped bar or as a soft curve.

Level: This controller pinches 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.

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. This 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. 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.

Sound cloner controls

Learn: A frequency analysis is calculated 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. This way the unedited signal can be compared to the result of the algorithm.

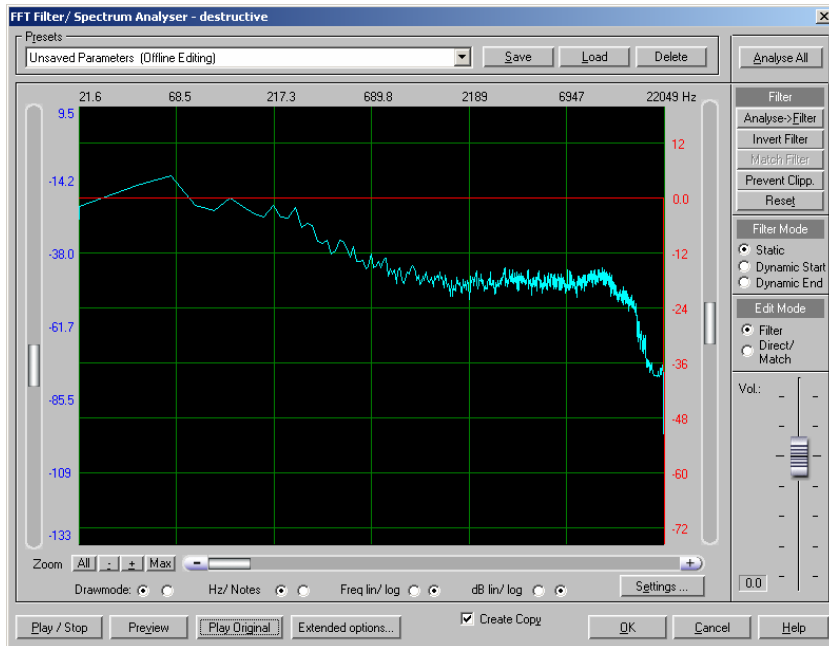
FFT Filter/Spectral Analysis (Offline)

With the FFT analysis filter you have a high-quality tool at your disposal which can be used to solve typical problems in elegant and partially completely new ways. The FFT analysis filter is a combination of a high-quality FFT spectral analyzer and an FFT filter. The calculated frequency response curve can be edited using a freehand tool and then sent to the audio material. The real-time preview function allows you to work in an intuitive and time-saving manner.

Some areas of application of this tool are:

- Checking and correction of the frequency response of the recording or mix.
- High-precision frequency measurement, e.g. for verifying the tuning of instruments.
- High-precision filtering of the finest nuances to full sample distortion without any changes to the phase response. Phase distortion effects like deteriorated impulse response or room impression are minimized with this. Linear-phase filters do not have their own sound.
- Sub-bass precision correction in the low-frequency range between 10 and 50 Hz with any increase or dampening without the effects that usually occur with filters of this kind. Ideal for bass-heavy pop music or for removing rumbling or footsteps.
- Selectively dampens or raises the level of individual instruments or pitches from recordings.
- Formant filter

- Dynamic filtering: Transition between two filter curves for filter switch effects, hiding frequency variables, and much more.
- Transmit frequency ranges: For example, from other mixes or rooms or loudspeakers to your own mix. This option can be very helpful if you want to improve the sound of your samples or simulate individual monitoring conditions.
- Correction of the frequency range from recording and playback areas, loudspeakers, amplifiers, etc.



FFT-Analysis filter - First step

To understand the basic operation of the FFT filter, we recommend that you now perform the following steps on your computer.

A small warning: reduce the playback volume before experimenting with the FFT filter as a precaution. With this technology you can easily increase the low frequencies to such an extent that your speakers may not be able to handle.

Note: To quickly understand dynamic filtering, we recommend using a sample containing a hissing sound. A corresponding sample can be quickly created using the Waveform Generator (view page 775).

1. Filtering and frequency analysis

Step 1: Select a range of a sample in your wave project or select an object in the arranger window of your virtual project.

Step 2: Open the FFT Analysis Filter again. A **blue curve** depicting the **current frequency response of your sample/object** is now visible. The **red curve** is the **frequency response filter**. By default it is set to 0 dB for all frequencies.

Under given circumstances, according to the length of the selected range, only a portion of the range is analyzed. To examine the entire range or object, click on the "Analyze all" button to the top right of the dialog window.

Step 3: Now use the left mouse button to draw in the graphic display. The frequency response of the filter changes according to the drawn curve. The blue curve also changes while drawing: it now shows the frequency course almost reached by the sample after application of the filter. The blue curve is therefore now referred to as the corrected frequency response.

Note: If you use a faster PC, you can check the result in real time after pressing the play/stop button. The changes may be heard immediately. Otherwise, you should use the non real-time preview ("Preview" button).

2. Filtering in Direct Draw mode

You can also draw the desired frequency response (the blue curve) with your mouse. Select "Direct" in "Draw" mode.

The **yellow curve** now displays the **original frequency response** as a comparison. Changing the frequency response in this mode is rarely needed and can lead to over modulation. For this reason, confirm that you want to "Prevent clipping" by clicking the button of the same name after every adjustment to the frequency response.

By editing the blue curve you will of course indirectly change the frequency response of the filter. To observe this, first reset the filter by pressing the "Reset" button. Switch back to the other "Draw" mode "Filter". Now you will see that the filter curve is set back to zero. If change back to "Direct draw" mode, edit the blue curve again and then switch once more back to "Filter" mode; you can see that the frequency response of the filter will have changed.

3. Dynamic Filter mode (fading between two filter curves).

Select the filter mode "Fade start". The displayed frequency response in this mode corresponds to the left border of the selected sample/object. You can also edit this frequency response in the Filter or Direct mode, as explained in the previous two points. However, filtering will now also directly affect the marked range of the sample or the selected object in the VIP.

Switch to "Fade end" and edit the frequency response the same way. The displayed frequency response in this mode now corresponds to the right border of the sample/object. In this case filtering affects the end of the marked range of the sample or the object selected in the VIP.

The red filter curve or blue curve with the corrected frequency response changes between both range limits and is faded from the first to second curve. Thereby the first curves fades into the second curve.

There are three different methods for fading available in the "Expert" dialog.

4. Filtering a sample/object with the frequency response of another sample/object

The procedure for this task can be divided into two basic steps: "learn process" and the filtering itself.

The learning process:

1. Mark a range of the sample/object of which you would like to use the filter curve for a second sample.
2. Open the FFT Analysis Filter again.
3. Press the "Analyze all" button.
4. Press the button "Analysis -> Filter".
5. Save the result as a preset.
6. Close the dialog by clicking on "Cancel".

You have now completed the "Learning process".

You can now apply the acquired filter curve to another sample/object.

The filter process:

1. Highlight the desired range of a sample/object to which you would like to apply the frequency response.
2. Open the FFT Analysis Filter again.
3. Load the saved frequency response by pressing the "Load" button.

Hint: For transferring frequency responses, you can also use the "Sound Cloner (view page 626)" feature integrated into the real-time FFT filter. The workflow in this case is simpler; however, you have fewer tools at your disposal.

5. Transferring the frequency response (spectral envelope) from one sample/object to another

This function is required, for instance, if the frequency response of a track mix (reference) is applied to another track. The filter curve is in this case calculated from the reference frequency response and the frequency response of the object that is to be changed.

The procedure for this task can be again divided into two basic steps: "learn process" and the filtering itself. To do this you should proceed as follows:

The learning process:

1. Highlight the desired range of the sample of which you would like to use the frequency response as a reference for the second sample. The selected range should have the highest "musical density", i.e. all involved instruments should be played. Drums or percussion should be available. The range length should not be under 30 seconds.
2. Open the FFT Analysis Filter again.

3. Press the "Analyze all" button.
4. Press the button "Analysis -> Filter". The filter curve now corresponds to the frequency response of the sample/object. Individual peaks indicate that the frequency response of individual tones has an influence on the result – in this case you should use a greater range for the analysis or smooth out the peaks with the freehand draw function.
5. Save the result as a preset.
6. Close the dialog by clicking on "Cancel".

You have now completed the learning process. It is now necessary that you compare the acquired reference filter curve with the frequency response of the sample/object in order to be able to calculate a filter function, which allows you to align the frequency response of the sample/object with the reference filter curve.

1. Highlight the desired range of a sample/object to which you would like to apply the frequency response. The selected range of the track should have the highest "musical density" here as well, i.e. all involved instruments should be played. Drums or percussion should be available. The range length should not be under 30 seconds.
2. Open the FFT Analysis Filter again.
3. Press the "Analyze all" button.
4. Open the saved reference frequency range by loading the corresponding preset.
5. Press the "Match filter" button.
6. If necessary, smoothen steep peaks or drops in the red filter curve by drawing along the red curve with the mouse.

FFT filter – graphical view

Overview of the curves

There are nine curves to select from in total. The yellow curve represents the original frequency response, the red one represents the filter curve, and the blue one represents the corrected frequency band. The blue curve therefore represents the frequency response of the spectrum after filtering.

In the "Filter" drawing mode, the red curve is edited, while in "Direct" drawing mode, the blue one is edited.

In the lower dialog of the "Expert" area, you'll have the chance to display all three curves, i.e. current frequency response, original frequency response and filter frequency response. To do this, select "3" in "Display -> Curve numbers" in the dialog area.

The first three-curve group is used in the "Static" filter mode, the second is the start setting "Fade start" for dynamic filtering, and the third in the end setting, i.e. "Fade end".

Faders and zoom buttons

Use the left fader to change the display range of the level of the yellow or blue curve, i.e. the original and the current frequency response. The level is shown in the left axis label of the graphic in dB.

- Use the fader on the right to change the filter range (red curve). If the fader is moved up, the value range increases in size. The frequency response can be increased to higher values or reduced to lower ones. This only involved fine-tuning and making corrections.

The smallest and largest values may be set in the "Max. range" and "Min. range" in the "Expert" dialog. The precise values for damping/dB increase are shown in the axis labeling to the right.

The volume fader moves the filter curve up or down and can be used to control the entire volume.

- The bottom left features four buttons for the zoom function. "+" changes to the next zoom level, "-" changes back a level. There are 10 zoom levels available in total. The "All" and "Max" buttons set the display to the smallest/highest zoom levels. Clicking on the "Max." button again will return to the previous zoom level. Use the fader beneath the display to move the curve section. The selected frequency range is then shown in the display in Hz.

Switch for the display views

Below the graphic, there are four switches provided to toggle between the different view modes.

Display mode: Displays the frequency response as a curve or line diagram. The line display provides a better overview in the low-frequency range, where there are fewer frequency ranges.

Hz/notes: This option allows you to display the graphic in Hz or musical notes. The note display is only available in the logarithmic frequency display (Freq: log).

Freq lin/log: This switches between logarithmic and linear frequency scaling. The logarithmic display corresponds to the human perception of volume. Like on a keyboard of a key instrument, the (graphic) spacing of the intervals, i.e. halftones, is uniform across the entire range.

dB lin/log: Select between a logarithmic or linear scale for the level. Logarithmic display corresponds with the human perception of volume.

FFT filter – controls

Filter

Analyze all: This calculates the frequency response for the entire sample/object. This function is only available in "Static" filter mode. Once this function is applied, the button deactivates.

Analysis > Filter: The red filter curve is calculated from the frequency response of the sample/object. Use the analysis to filter a sample/object with the frequency response of a different one or to transfer the frequency response to a different sample/object.

Invert filter: This inverts the filter curve. This function corrects the frequency range of rooms or loudspeakers.

Match filter: In this case, a filter curve is calculated that adapts the frequency response of the sample/object to the filter curve of a preset. This is useful if the filter curve contained in the preset has been acquired from the frequency response of a reference mix, since the sound characteristics of the reference mix may be transferred in a certain similar way.

Prevent clipping: If you edit the frequency response and the sample is overmodulated, the filter curve will be corrected correspondingly after pressing this button. This function is independent of volume faders.

Reset: Click on this button to reset all curves to their default (0 dB) position.

Filter mode

Static: In this mode, the edited filter curve is applied to the range of the sample in the wave project. When working in a virtual project, the edited filter curve is applied to the selected object in the virtual project.

Fade start: This mode provides dynamic filtering. The first curve (sample start) may be edited in this mode. The displayed frequency response in this mode now corresponds to the sample at the beginning of the selected range.

Fade end: Edit the second curve (sample end) with dynamic filtering here. The displayed frequency response corresponds with the sample at the end of the selected range. The resulting filter curve is now faded in between the two created curves. In the "Expert" dialog, three different filter fades are available.

Draw mode

Filter: Edit the red filter curve with the mouse in "Filter draw" mode.

Direct: This setting enables direct editing of the corrected frequency curve (blue curve). It is necessary for this setting to be saved in order to transfer the frequency responses of samples.

Expert: Opens a further dialog with additional setting options.

Play/start: Real-time preview function.

Preview: Non real-time preview function. In this case, full stereo operation is also possible. The result corresponds with the end product.

Play original: The unedited original sample/object may be played here for comparative purposes.

Help: Displays the "Help" feature.

OK: The filter is applied to the selected range of the wave project or the object selected in the VIP.

Cancel: The windows close without applying the filter.

FFT filter – expert dialog

Analysis parameters

The settings apply to the frequency analysis. They concern highly specific use-cases.

Analysis precision: During frequency analysis, the sample is split into individual blocks. On the "High" setting, the blocks overlap one another by 50%, while at "Normal", they simply

border one another. At very high overlapping levels, it's much easier to capture very short sounds, e.g. individual 1/32 notes. If you have selected "High", the calculation time for the analysis will double, which will become most noticeable when using the "Analyze all" function.

Analysis precision: In the "Noise" setting, analysis results for individual harmonics are slightly distorted and the peaks widen. With the "Sounds" setting the entire frequency response of noisy samples like drums, cymbals, or high hats may also be distorted a little. Inaccuracy occurs mostly at lower filter parameter rate values.

Channels: In case of stereo samples, select whether or not the right or left channel or both should be analyzed.

Analysis time: Specify how many seconds of the sample should be analyzed after 1.) opening the FFT Analysis Filter, 2.) when changing the setting for the rate, or 3.) after changing the settings for the analysis parameters. This setting only affects the display of the frequency response in "Static" filter mode.

Note: Every time a change is made to the settings of the analysis parameters or to the rate, the frequency response is recalculated. If you analyze the entire range of the sample using "Analyze all" before making changes to these settings, the results will be lost if these parameters are changed. In this case, select "Analyze all" again. The "Analyze all" button will be active once again in "Static" filter mode.

Filter parameters

These settings apply to the filter process.

Filter precision and real-time filter precision: This sets the quality of the precision. If precision is high, the calculation time also increases.

Low: The fastest method. In case of extreme filter settings in the bass area and at a low rate, low-level modulation noises will be audible. However, with less extreme filter settings, this setting functions without any noticeable loss in quality.

Normal: A practical compromise between CPU strain and audio quality.

High: Processing time increases by 30% more than the "Normal" setting. During dynamic filtering, fading is smoothest via this setting.

Max. range (dB), min. range (dB): Set the limit of the maximum increase or the maximum damping setting in dB for filtering. Fine-tuning may be applied using the fader on the right, beside the graphic display. If you want to hide individual parts of the spectrum, set "Min. range" to the lowest value (ca. -120 dB).

Threshold: When re-drawing the blue curve in "Direct" mode, only those frequencies that were originally louder than this threshold setting will be changed. This makes it easier to precisely increase or dampen harmonics.

Display

Apply additional settings for displaying the curve here.

Grid: Switches the grid on and off.

Number of curves: Set the number of curves to be displayed here.

Filter transition

There are three different modes available for dynamic filtering:

Direct: A simple fade is made directly between two curves.

Move/direct: In this case, both curves are faded, but the start curve is to the side, i.e. it is moved onto the frequency axis. For example, if a band pass is set for the start curve, the timing of the mid frequency changes (filter sweep). The corresponding high value of the start and end curve specifies the extent of the curve move. If you are working in "Filter" draw mode, edit the red curve first. If the entire filter curve is set to 0 dB, a higher value will not be available.

Move: This method can be used to move the start filter curve. Once again, the corresponding high value of the start and end curve specifies the extent of the curve move. The second curve is used for specifying the "end point" of the move. Otherwise, it doesn't have any influence on the result.

Resolution: This parameter affects the analysis and filtering processes equally. Set how many individual analysis frequencies/filter bands should be available, in which case the number of filter bands corresponds to exactly half of the value of the "resolution" parameter. The higher the resolution, the longer the processing time needed – by approx. 5%/level.

FFT - Filter - The right settings**Frequency analysis**

A higher resolution is required for sufficient precision in the low-frequency ranges (from 8192). The precision of the analysis in Hz at a sample rate of 44.1 kHz can be found in the table.

Resolution	Precision in Hz
256	172
512	86
1024	43
2048	21.5
4096	10.7
8192	5.3
16384	2.7
32768	1.35

Filters

Here you only need high resolution (8192) for low frequencies or for specifically blocking out overtones. For all uses the settings 1024 to 2048 will suffice. Values lower than 1024 are rarely needed.

Notes:

If you have drawn very detailed filter curves, it is imperative that you save the curve parameters. When switching modes or parameters, the curves will appear to be closely matched but will not maintain the same details.

Make sure that when previewing in realtime at a high rate that the value of the program buffer in the ("**Extended Buffer Settings**" (view page 500)) is at least just as high, or even better, a bit higher than the set rate. Otherwise real time monitoring is not possible. Rates higher than 8192 are hardly useful when previewing in real time, since the internal real-time buffer size of the algorithm corresponds to the set rate. In this case, editing while previewing in real time will be "awkward".

Save/Load: Here you can save or load all parameters and curves. The default file extension is *.fff.

Important: To apply the **filter to other samples**, the "Draw" mode parameter has to be set to "Filter". Only in this case will the red curve be saved. This also applies if the filter is indirectly applied by editing the blue curve in "Direct draw" mode.

However, if you save when "Direct" is selected, **the saved frequency range may be applied to other samples**. In this case the saved blue curve is transformed into the red one when loading.

FFT Filter - Problems and solutions**Modulation effects when using extreme filter settings for low frequency ranges**

Set the "Precision" setting in the "Expert" dialog from "Low" to "Normal" or "High".

Problems with the realtime preview when using higher rates:

Make sure that when previewing in realtime at a high rate that the value of the program buffer in the ("**Extended Buffer Settings**" (view page 500)) is at least just as high, or even better, a bit higher than the set rate.

It is not possible to completely hide certain areas of the spectrum.

Select "Filter" in draw mode and reduce the "Min. range" value in the "Expert" dialog to approx. -100 / -120 dB.

You have corrected the frequency response of a sample and then analyze the result once again. The result does not match the corrected frequency response (the blue line before performing filtering).

If you use the "Analyze all" function and set the filter parameter to "Precision: High" in the "Expert" dialog when analyzing before and after filtering, an approximate match will be set.

Also resonance-like sounds arise in wideband band passes.

Reduce the slew rate of the band passes by not creating any vertical lines while drawing.

In the "Fade start" or "Fade end" filter mode, fading is not applied uniformly.

In this case, select the highest setting for the filter parameter precision in the "Expert" dialog. In "Direct" fade mode, the problem is reduced when the rate is reduced. Experiment in the other blend modes with this parameter to find a suitable setting.

Brilliance Enhancer

(Optional as part of the Cleaning/Restoration Suite)

You can find more information on the Cleaning/Restoration Suite in the PDF document "Cleaning/Restoration Suite" included with the program.

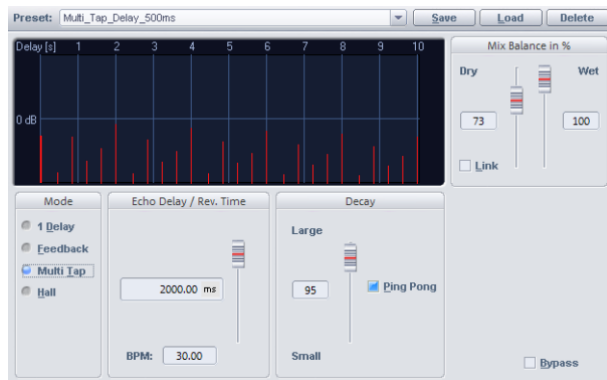
Filtox

More information is available in "Effects -> MAGIX plug-ins (view page 698)".

Delay/Reverb

Delay

Use this function to integrate an echo effect into a sample/object. To avoid overmodulation, you should set up a certain modulation reserve.



Echo delay/reverb time: This controller sets the delay between the individual echoes or the original signal and the first echo in milliseconds. Note that the delay is dependent on the sample rate, i.e. a change in the sample rate after creating the echo leads (logically) to a change in the echo delay.

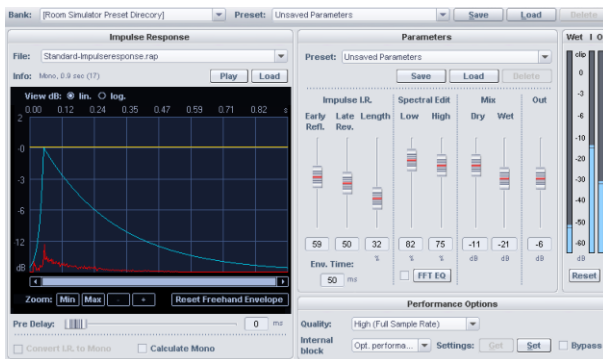
Decay: Set the decay value for the simulation of the size of the room numerically as well as using faders. In "Offline" mode, there is also a "ping pong" effect available, and you can also set the reverb chamber in the real time dialog.

Mix balance in %: Enter the ratio of the original signal to the echo/reverb signal in percent here. If you switch on the "Link" option, both values always add up to 100%.

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 sample. A room impulse response is the reverb of an 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 - general controls

Room Simulator - banks and presets

The Room Simulator offers several reverb banks. These "banks" divided into categories like "ambience", "drums_percussion", or "small reverb", and these contain the corresponding presets and impulse responses.

Loading a preset loads both the parameters of the dialog field and the impulse response. The impulse response may be any sample saved to the hard disk as a WAV, HDP, or RAP file. The impulse responses that come with the program have a special file format with an *.IMR extension.

If a file has been deleted or moved in the meanwhile, an error warning will appear when loading a preset. Correspondingly, when a preset is being saved, the selected impulse response itself will not be saved, but only its path and file name. Presets have the extension *.rms. Files with impulse responses from the program presets are located in the "RoomSim" directory or in the bank directories.

The presets can be saved, loaded, or deleted. The load function is only needed if the preset files are located in a different directory than the "RoomSim" directory.

Room Simulator saving/loading/deleting presets

Presets can be loaded, saved, or deleted.

The presets that are located in the "FX-preset" subdirectory are automatically loaded and listed in the dropdown menu, where they can be selected. The "Load" function loads presets from directories besides "RoomSim" into the Room Simulator.

Impulse response

File: Select an impulse response here. Impulse responses may be loaded from the "RoomSim" directory, the bank directory, and from the wave projects opened in the program (Bank: [loaded projects]).

In case during Samplitude installation you have deactivated the "Install impulse response" option, you will have access only to the opened wave projects (Bank: [loaded projects]) and "Standard-impulse-response.rap" from the "Room Simulator preset directory" bank.

Info: This provides information about the length of the impulse response and whether it will play back in mono or stereo.

Play (impulse response)

To verify the impulse response, this is played after this button is activated. 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 in any directory as a WAV, HDP, or RAP file.

Graphic display

dB lin: The graphic display with the impulse response has linear amplitude scaling. This setting corresponds to the usual sample representation.

dB log: Choosing this setting will scale the amplitude display of the impulse response logarithmically. This corresponds with the way the human ear perceives volume changes.

Zoom options (all, in, out, max, position):

These options zoom into the display to select the range to be displayed.

Curves

Red: Impulse response timeline

Light blue: 2-segment envelope curve to suppress early reflection and post-reverb.

Yellow: Freely definable envelope curve

Freely definable envelope curve

The amplitude progression of the impulse response can be manipulated with the yellow freely-definable envelope curve. Typically, it is used to dampen or delete early reverberations of the impulse response. Using zoom options will aid greatly in this task.

Reset freehand envelope: Reinstates the curve's original status.

Pre-delay

The "Pre-delay" function delays the entire impulse response by a time interval variable between 1 and 100 ms.

Convert impulse response to mono

If you choose this option, stereo impulse responses will be recalculated in mono. The convolution count remains unchanged in stereo (it corresponds to the count of group channels in case of Surround).

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. In the case of Surround sound, all channels of a group are added to a 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 lowers or suppresses early reflections by dampening the first part of the impulse response.

Late reverb: This parameter decreases or suppresses 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. Note that reverb may end rather abruptly, which may lead to unnatural decay patterns. Compensate by applying a lower late reverb setting to fade the impulse response with the 2 segment envelope curve. You can optically control the impulse response envelopes on the graphic display.

Crossover: Set the time length of the first of the two segments of the envelope curve here. This parameter influences the dampening of the earlier reflections.

Parameters for editing reverb frequencies

Lows: This parameter adjusts the low frequency component of the reverb.

Highs: This parameter adjusts the high frequency component of the reverb.

FFT EQ: The reverb component can be edited with an additional **FFT filter**. The graphic display includes a real-time spectrum of the reverb component in the FFT filter to provide visual control of the frequency characteristics of the room in which the impulse response was recorded. This enables unwanted responses to be located and removed quickly.

On lower settings (< 4096 samples), the FFT filter is not available for the latency parameter. This is because its linear function creates increased latency.

Mix, volume, reset

Original: Set the original signal level in dB here.

Reverb: Set the reverb level in dB here.

Volume: Set the master output level in dB here.

Reset: Set the level to its original position here.

Performance/options

Quality: In both "normal" modes, the room stimulation will be calculated using only half the sampling rate. In most cases, this will be quite enough, since natural impulse responses and even impulse responses generated by digital reverb devices rarely possess components above 10 kHz (check this in the spectral representation of the integrated FFT filter). Many other reverb devices work internally with the half-rate, since calculation over the entire frequency range would be a waste of CPU capacity.

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.

"High" mode calculates the entire frequency scale. The CPU load doubles in comparison to "Normal" mode.

Internal block length: This parameter specifies a block length to calculate the convolution function internally.

Short block lengths raise the count of the required calculation operations, causing CPU strain to increase. Large block lengths lead to an uneven CPU loads. The parameter has no effect on the calculation result itself.

The optimum for real-time processing tends to be about 16384 to 32768 samples.

For use in the AUX, track, or master without low latency, a setting of 8192 or lower is recommended on faster systems (Pentium 1800 or Athlon XP 1500+ and higher) for an acceptable response latency.

For use in low-latency conditions, the value can be reduced to 32 on faster systems (from Pentium 2800).

The value "optimal latency" adjusts the internal block length to keep the latency as low as possible without influencing the performance too much.

The value "optimal latency" adjusts the internal block length to keep the performance as high as possible without raising the latency too much.

Note: Setting a value smaller than the set buffer size is not useful. If the value corresponds to the ASIO buffer size, the Room Simulator processing is latency-free (only with "higher quality").

The FFT filter cannot be used at latencies less than or equal to "4096". Homemade impulse responses provide the opportunity to use the FFT filter destructively on the impulse response before calculating the room simulation.

The block length is raised internally when using the Room Simulator for destructive processing, since processing at a lower latency would result in an avoidable increase of the necessary CPU operations.

Retrieve, set: Saves and sets the quality options globally. Note that when using this on the object and in the track, different settings may be applied for AUX insert effects or master.

Bypass: The "Bypass" button plays the original signal.

Overview of the impulse response parameter

Quality

"High" setting: Doubles the quality

Internal block length

Low internal buffer sizes (2048 samples): Increases the internal block length by approx. 50 percent.

Very low internal buffer size (128 samples): Increase in the internal block length by approx. 150-200 percent.

Length of the impulse response

If the length of the sample used as an impulse response exceeds a power-of-two, the CPU load doubles.

The performance required subsequently doubles whenever the following impulse response lengths are exceeded:

11.88 s
 5.944 s
 2.97 s
 1.48 s
 0.74 s
 0.37 s
 etc.

Note: Make sure that homemade impulse responses are shorter than the next value, particularly if the impulse responses are longer than 1 second, since the length of these "continuous CPU load intervals" increases exponentially as length increases.

Sample rate of impulse reply

The CPU load increases proportionally to the sample rate.

Impulse response resolution

The CPU load is independent of this, since computation is carried out with float precision in each case.

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 with the impulse response enables quick optical control 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 672)"

Heavy hard disk activity or an error message concerning insufficient memory was received.

This algorithm requires a lot of memory, especially for long impulse response patterns. All background processes should be terminated and all RAM wave projects that are not needed should be closed.

CPU power usage is too high.

- Check the "Internal block length" parameter's setting in the parameter's description.
- Activate the option "Quality: Normal".
- Shorten the length of the impulse response. Note the information provided above regarding intervals with the same CPU strain.

Problems with very long impulse responses.

Real-time operation on modern computers: Impulse responses of up to 11.88 seconds may be applied (with 44.1 kHz sample rate). The length is only limited by the CPU capacity. For destructive calculation, the upper limit is 380.4 seconds (44.1 kHz sample rate). If this upper limit is exceeded, the samples behind will be ignored during calculation. Impulse responses of this length are, however, not practical for room simulation. When experimenting with very long impulse responses (longer than 1 minute), note that the algorithm usually requires a lot more RAM (over 500 MB). When loading long wave projects as impulse responses, a warning appears to indicate that the calculation may lead to long waiting periods.

eFX_Reverb

Detailed information on eFX_Reverb can be found at "essential FX > Reverb (view page 687)".

eFX_StereoDelay

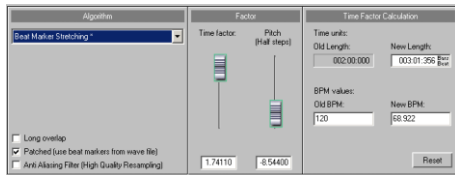
Detailed information about StereoDelay is available in "essentialFX > StereoDelay (view page 689)".

Ecox/Variverb

More information is available in "Effects -> MAGIX plug-ins (view page 698)".

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 682)".

Resampling

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.

This effect is relatively loss-free; the sound loss is less than with Time Stretching/Pitch Shifting procedures. If a pitch change is desired or justified by changing the speed, then 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 571) ("Object" menu).

Default

In this case, an algorithm is applied that usually delivers very good results with factors from 0.9 to 1.1, that works in phase-locked mode, and therefore maintains the spatial effect of stereo signals. This algorithm is only partially suitable for drum loops or other "beat heavy" audio material since it may affect the "groove".

Time compression (sample length is reduced) is more successful with this algorithm than Time Stretching, i.e. it is better to reduce the longer sample than vice versa.

This algorithm is especially suited to complex audio material. The CPU load is light.

On extreme settings, aliasing effects may occur. An additional anti-aliasing filter may help in this case. This filter is also available for all other algorithms that work internally with resampling.

Beat marker stretching (smoothed)

Beat marker stretching (smoothed) represents the standard methods for high quality Time Stretching and Pitch Shifting. This algorithm works basically the same way as "beat marker stretching". In this case, the stretched material is synchronized with the beat markers so that the groove remains perfectly intact. Furthermore, the smoothed algorithm is used in between the beat markers. With percussive audio material, good audio quality is possible even at very large time factors. The CPU power usage is also increased in this case.

Smoothed

This deals with a complex algorithm that requires higher processing time. The audio material may now also be used at very large factors (0.5 – 2.0) without causing severe artifacts. The audio material is "smoothed", which makes the sound softer and produces an adjusted phase level. This smoothing is hardly audible, e.g. with speech, singing, or solo instrumentation. Problems in the shape of distortions may result with more complex spectra (sound mixes from various instruments or finished mixes). This algorithm is less suited to drum loops and other audio material featuring strong transients. The groove will remain intact, but phase shifting may "smudge" the attack. At smaller corrections (factor approx. 0.9 - 1.1), the smoothing degree setting of "very low" is often sufficient.

The "smoothed" algorithm is especially useful for:

- Orchestra instruments, string instruments, wind instruments
- Monophonic and polyphonic speech
- Speech recordings featuring background noise (e.g. video recordings)
- Synthesizer areas, guitars

The "smoothed" algorithm is less useful for:

- Stereo mix
- Drum loops, percussion

CPU power usage is too high.

Beat marker slicing

This mode is especially useful for customizing drum loops, but it may also be used on other audio material like monophonic bass runs or sequencer lines. The algorithm splits the material into individual components of notes or beats, which are then selected via the beat marker.

These "audio snippets" are then compiled into the new temporal measure: if the tempo is increased, individual beats begin overlapping; if it is slowed down, short pauses between the beats become audible.

Use this algorithm whenever possible, since timeshifting is possible in this case without having to sacrifice quality and at minimum CPU usage.

The algorithm **"Beat marker slicing"** is especially useful for:

- Samples that are able to be split into individual notes or beats
- Drum loops, provided the beats do not overlap and there is not much reverb

The algorithm **"Beat marker slicing"** is less useful for:

- Signals that cannot feature beat markers. Silence will be the result.

The CPU load is very light.

Beat marker stretching

This algorithm works in a similar way to the standard algorithm, the main difference is that the stretched material is synchronized with the beat markers. This way, the groove remains perfectly intact and doubled or faded out beats will not occur. Unlike the "Beat marker slicing" algorithm, absolute silence does not occur when there are no markers or incorrectly placed markers; instead, the synchronization quality is reduced or has errors. In practice, automatic marker generation is usually sufficient for good results. Avoid short distances with sequential beat markers (< 1000 samples).

Long overlap

If this option is active for "Beat marker stretching", the audio will not be as rough, however echoes may result. This option is especially useful for drum loops featuring slight time changes.

The **"Beat marker stretching"** algorithm is especially useful for:

- All types of rhythmic samples that are unable to be cut into individual notes or measures (beats)
- Rhythmic samples of all types, although individual notes/hits will not be separated
- Applications that cannot be used for beat marker slicing or are too complicated for perfect beat marker positioning.

Setting and evaluating beat markers

If beat markers have not been placed yet, these will be generated automatically in real time. In the Resampling/Time Stretching dialog, select the **"Patched"** option to specify whether or not saved ("patched") beat markers should be used in the wave project.

Use the following option to generate beat markers:

Find and use beat markers: After activating this option, the algorithm will locate the beats and mark them as beat markers.

Note: This function is only available in the wave editor.

Attack recognition threshold

Use this fader to set a threshold for beat recognition. When the wave project displayed, you can follow the placement of beats visually. This also works with VIP objects if the wave project is open at the same time.

Reset Beat-Marker: All Beat markers are removed.

Beat markers can be corrected manually. Beat markers can be re-positioned, copied, deleted etc. by activating the applicable marker commands.

Hint: If you wish to set new beat markers manually, right-click on the marker bar and select "Save marker -> Markers with names...". Name your new marker as "**Beat**".

We encourage you to set the value of the velocity detection threshold to a high value before editing and then to delete the superfluous markers - this is far quicker than trying to expand missing markers manually.

With the Timestretch/Pitchshift Patcher (view page 571) in the "Object" menu you can easily set the markers. This give you the option to edit the Beat marker, if the dialog is open (non-modal window). Thus you can check the fruits of your labors immediately, without having to close the dialog. Moreover in the patcher, the tempo (BPM) and Timestretching mode used by the object editor can be saved directly in the wave project.

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.

Universal HQ

This algorithm provides a high-quality Time Stretching/Pitch Shifting method and offers good audio quality with almost any audio material. Especially when it comes to complex audio recordings like orchestral recordings, this algorithm delivers high-quality results. Stereo properties also remain intact.

The CPU load can become very high when realtime effects are applied to many simultaneously playing objects.

True Resynthesis Timestretching (TRTS)

How does True Resynthesis Timestretching (TRTS) work?

The signal is broken down into individual components, with the amplitude and frequency modulated oscillators - simple synthesizers -are reproduced faithfully. If these synths play longer or shorter a Timestretching effect will be created.

What is the difference to traditional TS algorithms?

All traditional algorithms work on the basis of signal-adjusted, automated cut functions (Copy, Cut, Fade etc).

With TRTS however you can stretch timelines when "Resampling". The typical compression artifacts and associated non-uniformities will not occur.

Will Artifacts emerge?

The artifacts are small in comparison to conventional processes. Linear stretching impulses, sibilants and wide band noises leads to a change in sound characteristics and can be produce artifacts. Tonal components can be stretch perfectly.

Are beat markers used?

Yes. At recognized beat markers the stretch factor is internally set to 1, so that the beat isn't affected.

What advantages does the algorithm offer?

- Universally usable. Due to linear work practices TRTS is particularly suitable for very small and very large Timestretch factors.
- Detected transients remain true to the original, undetected transients are only slightly changed.
- It can also be used for material with complex rhythms (Jazz, Salsa etc). Here conventional algorithms have more difficulty than in straight disco or techno beats.
- Particularly well suited to orchestras.
- No trimming or filtering out of signal components - thick, rich sound.

Which audio types could cause problems?

- Speech, especially deep male voices
- Sibilants

- Computer-heavy pop and techno music, due to the discoloration of noisy sections.

How high is the CPU usage? How much memory is required?

Multiple CPU cores are used. On faster systems (Intel i7 - Processor 3 GHz) the algorithm calculates stereo samples in near realtime at a sample rate of 44.1.

Memory requirement is dependent on the material and is somewhere in the region of 200 MB.

How will stereo behavior and spatial qualities be influenced?

The stereo position of components remains the same. Since the reverb portion is either lengthened or shortened when stretch, the room size will also be correspondingly be increased or decreased.

Elastic Audio

Shortcut: Ctrl + Shift + E

Elastic Audio – general

Elastic Audio is a specialized editor for changing the pitch of audio material. A combination of automatic resampling and Pitch Shift Algorithms as well as pitch detection for monophonic material allows extensive changes to be made to the pitch for audio material.

Pitch Shifting Algorithms are able to change a melody in the pitch without influencing the tempo. Samplitude provides a variety of Pitch Shifting Algorithms, which can be used depending on the existing 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.

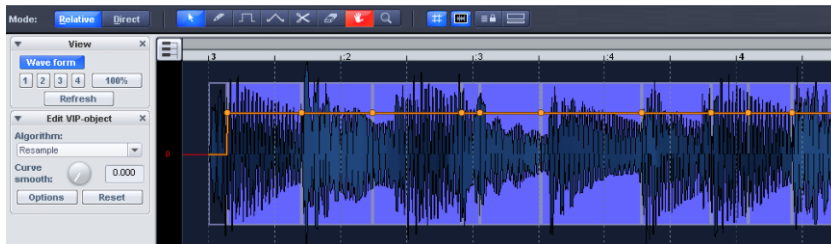
Note concerning Time Stretching: Time Stretching cannot be automated, but it can be used statically with Elastic Audio.

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.

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

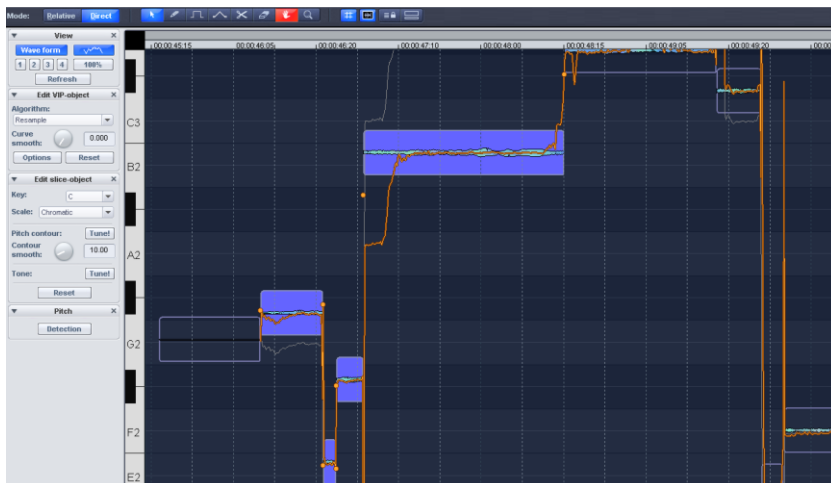
Y-axis: Displays Pitch Shifting in semitone steps (within a range of -48 to 64 semitones).

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

Red 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.

Elastic Audio – basics

Opening Elastic Audio Editor

Select the objects to be processed with Elastic Audio Editor. The objects may be located in several tracks. Elastic Editor is launched via "Effects" or the Time Stretching section in the object editor.

Shortcut: Ctrl + Shift + E

Interaction between Elastic Audio Editor and objects in the arranger window

First, make temporal corrections to notes by splitting and moving objects in the arranger window.

Elastic Audio Editor is multi-object capable, which means it can edit a melody that has been cut into individual objects as a whole.

Loading objects into the Elastic Audio Editor

Elastic Audio Editor loads all objects that were previously selected. However, you can only ever view objects from one track and edit them simultaneously. When objects are loaded into Elastic Audio Editor from several tracks, switch the track displayed in the editor via "Edit VIP objects" via the track selection.

Elastic Audio: Description of all control elements

The control elements available in each mode are available for activation/deactivation in the lower section of Elastic Audio Editor's "View" menu.

View



Waveform: This turns the waveform display on and off.

Zoom buttons: Save the zoom depth and position for the current window here. Buttons 1-4 or numbers 4, 5, and 6 of the number block are available for this purpose. These also access the zoom buttons 1, 2, and 3 respectively. The fourth zoom button is not assigned to any key in the number block.

Save Zoom Level:

Shortcut: Ctrl + number block 4

Ctrl + number block 5

Ctrl + number block 6

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

Get Zoom Level:

Shortcut: Number block 4

Number block 5

Number block 6

"Shift" + left click on the zoom button will overwrite the last zoom level.

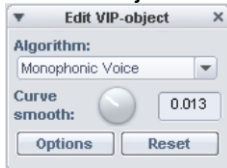
"Ctrl" + left click opens only the horizontal position of the corresponding zoom level.

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

Note: The key combination "Ctrl + Alt" + mouse wheel enables simultaneous horizontal and vertical zooming.

Refresh: This refreshes the graphical display of the material. Any resulting changes to the length are adjusted as such. In case synchronization of the arranger and Elastic Audio Editor is active (Elastic Audio Editor -> Display -> Horizontal -> Synch. VIP editor), then the effects will be displayed in the VIP within the arranger window.

Edit VIP object



Track selection: If objects from multiple tracks have been loaded into Elastic Audio Editor, a track may be selected here.

Algorithm: Select between "Resample" (preset), "Monophonic Voice", "Standard", "Smooth", and "Beat Marker Slicing" modes.

Detailed information about these algorithms is available in "Effects -> Time/Pitch -> Resampling/Time Stretching/Pitch Shifting (view page 647)".

Smooth: This parameter smoothes the pitch curves entered across a time constant (in ms). Peaks in pitch changes become a glissando in case of stronger smoothing.

Options: These are advanced settings that depend on the selected algorithms.

Reset: Undoes all changes to automation curves or voices of selected objects.

Options for automatic pitch correction

Edit slice object (direct mode)



Key: This option sets the basic key of the scale. In the chromatic key, this setting is not included in calculation.

Scale: Use this option to select the type of scale, i.e. the sound make-up/mode. Choose from "major", "minor", "harmonic minor", "pentatonic", and "chromatic".

Pitch contour: Tune!: This button quantizes ("levels out") the pitch contour of the selected slices.

Contour smooth: Set the strength of the quantization (lower values quantize hard). This function removes repeated small pitch alternations in natural sound sources, e.g. vibrato ("Cher" effect).

Tone: Quantize the mid pitch of the selected slices. The corresponding sections of the pitch contour of a slice object are moved together and adapted to the mid pitch. Pitch fluctuations within a slice object remain the same.

Note: Automatic correction requires the slice's pitch (pitch -> detection).

The results of the calculation of mid pitches for slices featuring glissandos often do not correspond to the pitches that human hearing assigns to them. In this case, automatic correction of the mid pitches does not give satisfactory results. Removal of the glissando passage by manually changing the slice borders (keyboard shortcut: Ctrl + rubber handle tool) or the splitting of slices can help.

Reset: This command resets the selected slices. The orange curve is superimposed over the gray curve and the slice is reset to the original mid pitch.

Harmonizer

Use the **harmonizer** to add additional voices to the object. Please note that the harmonizer is only available in "Direct" mode via the "Monophonic voice" algorithm. Format correction is required for a satisfactory result which is only possible with this algorithm.

Chord: Select the chord created by the voices if the "Harmonic adjustment" parameter has been selected.

If "Harmonic adjustment" is deselected anyway, the voices are sent so that they are overlaid onto the sounds set in "Edit slice object". The set chord then only serves as a "guideline".

Humanize A "human touch" may be applied by moving the individual voices more or less subtly away from each other.

Harmonic adjustment: Activating this field causes voices to run parallel to one another and correspond to the set chord.

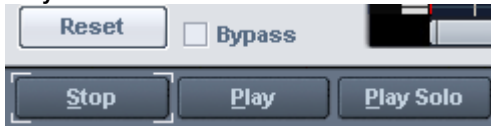
Create voices: Use this button to recreate parallel voices.

Pitch (fundamental frequency)

Detection: Use this button to start the basic frequency analysis. In this case, the absolute pitches are displayed graphically in the Elastic Audio Editor. This function is indispensable for additional editing steps you may wish to make in "Direct" mode. For this reason, you

should only open the pitch recognition before executing pitch corrections or other editing processes.

Playback control



Reset: All pitch curves for the entire material loaded into the Elastic Audio Editor are reset.

Bypass: Plays the arrangement without editing.

Stop/play: Stops/starts the playback of the arrangement.

Play solo: Only activated objects in the editor are played.

Toolbar

Several tools are available for processing the slices and pitch envelope. Two tools can each be assigned to one of the two mouse buttons. The tool assigned to the left mouse button turns blue; the function assigned to the right key is marked in red. Click with the corresponding mouse button on the desired console button. The "Zoom" tool is an exception: both buttons are assigned automatically.



Selection tool (arrow): Use this tool to move slice objects vertically. This modifies the pitch of slice objects as a whole. Slice objects and curve handles may also be selected, and a slice object lasso can be dragged out. Multiple selections are possible with the "Ctrl" or "Shift" key.

Freehand drawing function: Use the pencil tool to draw the pitch curve freely. If the "Shift" key is pressed, a straight line is drawn from the starting position to the current position of the mouse. When the "Ctrl" key is also pressed, the slice objects are combined during drawing.

Pencil tool for quantized drawing: This is the tool for quantized drawing. Quantizing means that the line is free horizontally, but snaps into semitone steps vertically in "Relative" mode, and to the tones of the scale selected via "tune" in "Direct" mode.

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

Rubber handle tool: This rubber band tool may be used to bend the pitch envelope between two neighboring handles. The range between the curve handles is moved, but the handles themselves remain unchanged. The curve is bent inwards at the handle point. In combination with moving the curve handles on the slice object's boundaries, you can bend the tone envelope and thereby maintain the microtonal structures (vibrato).

Cut: Use this tool to manually split the audio material into slice objects. After automatic pitch recognition has been applied, use the "Cut" tool to split certain passages into

individual slices. To recompile slice objects, use the pencil tool to drag along the slice while holding down the "Ctrl" key.

Eraser: The orange curves are set back to their initial values, i.e. the pitch envelope corresponds to its original curve again.

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

Zoom tool: Click with the left mouse button to zoom into the display; zoom out with the right mouse button. Left clicking and dragging stretches the range displayed as the zoom area.

Controls for global display options

These buttons have a global effect for the Elastic Audio Editor.



Grid: Use this function to switch the raster on or off.

"Single object" mode: The maximum horizontal zoom level is limited to one object.

"Synchronization" mode for VIP and Editor: Horizontal zoom levels of both windows are synchronized with each other.

"Docked view" mode: When "Docked view" is active, the arranger window is located below the Elastic Audio Editor.

Using Elastic Audio Editor

Pitch slice objects and VIP objects

During the **basic frequency analysis**, the objects in the pitch slices are split, which ideally corresponds with a note of a melody in monophonic music or monosyllabic speech. For speech, the pitches correspond with the positions of the vocals. After analysis, all slices are displayed in blue.

The splitting of slices can be corrected manually. Slices are split using the cut scissors tool and glued together using the rubber handle tool while holding down the "Ctrl" key. The blue line displays the medium pitch for every object, which is the approximate result of the average value of the pitches in the slice.

The functions contained in "Edit slice object" may be used on **selected slice objects**:

- Pitch Contour Quantization
- Tone Quantization

After pressing the "Tune!" button, the changed setting will be audible. Pressing "Reset" undoes corrections made to the selected slices.

The functions contained in "Edit VIP object" may be used on **selected VIP objects**. This involves selecting an algorithm and the smoothing magnitude for the pitch or automation curves. Pressing "Reset" undoes corrections made to the selected slices.

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

Selecting slice objects and VIP objects

Slice objects can be selected/deselected by clicking them with the mouse. The keyboard shortcut "Ctrl + A" selects all slices; de-selection is made by clicking on "empty space".

VIP Objects cannot be selected directly in the Elastic Audio Editor. When a slice object is selected, the corresponding VIP object is also selected.

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 as previously explained in the resampling section. A pitch algorithm may also be selected (standard, smooth, monophonic voice, beat marker slicing). The length of the VIP objects remains the same here.

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

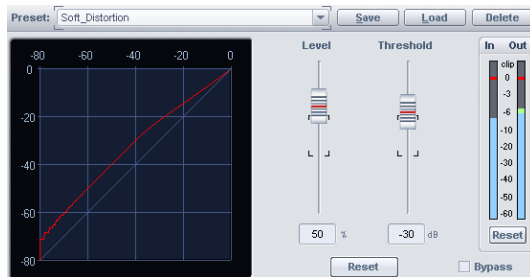
- Create a duplicate of the object on the track below it in the arranger
- Selection of both objects in the VIP
- Open Elastic Audio Editor
- Select the track via "Edit VIP object -> Select track"
- Edit each track individually. Do not use the resampling algorithm to avoid different tempos in the voices.

Alternatively, additional voices may be created easily with the "Harmonizer" in "Direct" mode.

Distortion

Distortion

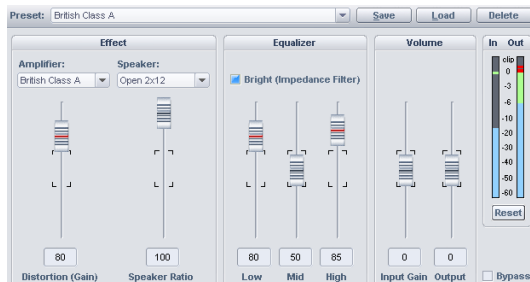
This effect allows you to distort audio material by using a non-linear transfer reference line. The signal becomes louder and overtones are added.



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".

Amp simulation

This menu item is hidden by default. To display it again, activate it in the menu settings via "File -> Program Preferences > Edit Keyboard Shortcuts and Menu (view page 488)".



The Amp simulator is a realtime effect that simulates tube amp sound properties. It is especially suitable for guitar sounds, but also for the organ or recordings of vocals.

Presets: Here predefined amplifier simulations can be selected.

Amp: Here you can select from a range of amplifiers whose sound can be simulated.

Speakers: Here you can select from a range of speakers whose sound can be simulated.

Distortion (gain): This regulates the distortion amplification.

Speaker component: Here you can regulate the speaker component present in the complete signal.

Equalizer: A 3-band equalizer for setting the bass, mids and highs.

Bright (impedence filter): he "Bright" button provides an especially penetrative acoustic pattern that sounds like screeching when combined with the input gain.

Input gain: With this fader you can adjust the volume of the distortion.

Output: Here you can adjust the master volume of the amplifier module.

Vandal

Detailed information about Vandal is available in "Vandal (view page 745)".

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.

Restoration

In Samplitude the following restoration tools are available as standard:

- DeClipper SE
- DeClicker/DeCrackler SE
- DeHisser SE
- DeNoiser SE

Furthermore, you can remove noise from an object with the mouse mode "Spectral mode (view page 72)". Editing can be done directly in the Arranger window.

With the optional "Cleaning/Restoration Suite" you have access to the full versions with an enhanced range of functions.

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.



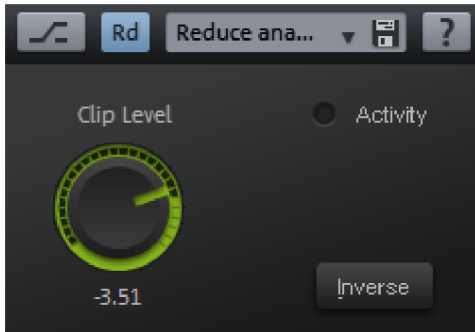
By clicking on this button you can toggle between the selected automation write/read mode. By right clicking you can get to the Automation context menu.

Via "?" you can call up extra help information.

DeClipper SE

(The full version is available as an option of part of the Cleaning/Restoration Suite)

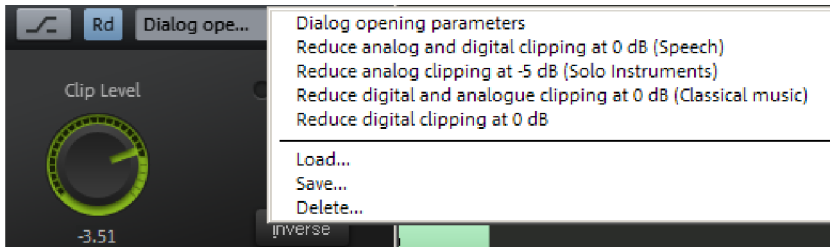
The De-clipper is a tool for removing overmodulation and distortions. Overmodulated passages are recalculated, or interpolated, based on the material immediately surrounding them.



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.

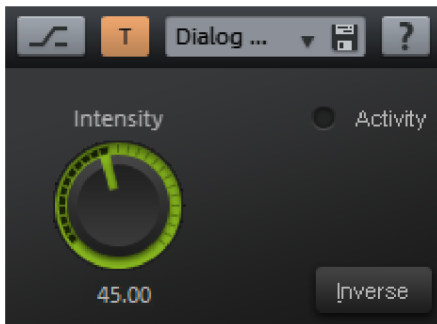
Activity: This LED flashes blue if the signal is altered using the effect.

Inverse: If this switch is activated, you will hear only the part of the signal that is changed by the algorithm.

DeClicker/DeCrackler SE

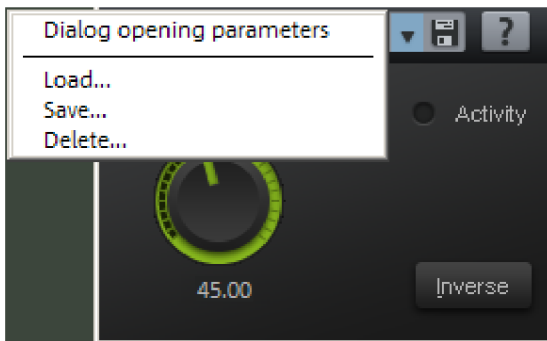
(The full version is available as an option of part of the Cleaning/Restoration Suite)

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.



De-clicker/De-crackler 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.

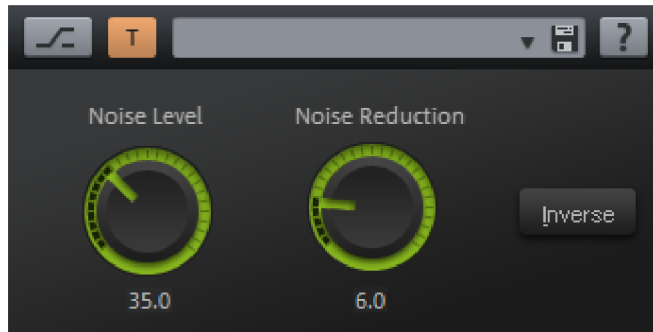
Inverse: If this switch is activated, you will hear only the part of the signal that is removed by the algorithm. If the parameter is optimally tuned you will only hear the bothersome

clicks. If the parameters are too high, parts of the music or speech signal will also be filtered, which will lead to music discoloration.

DeHisser SE

(The full version is available as an option of part of the Cleaning/Restoration Suite)

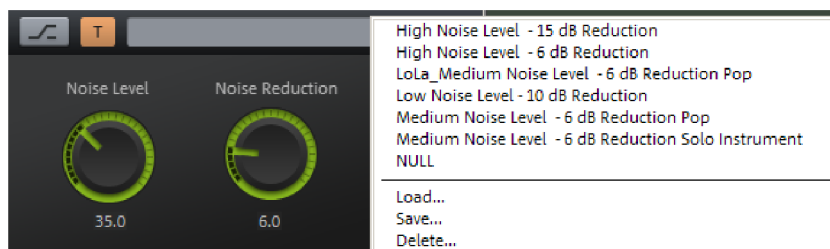
This command opens the DeHisser SE, which helps to remove noise. The DeHisser SE eliminates regular, low-level "white" noise, typically produced by recordings, microphones, pre-amplifiers or transformers.



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.

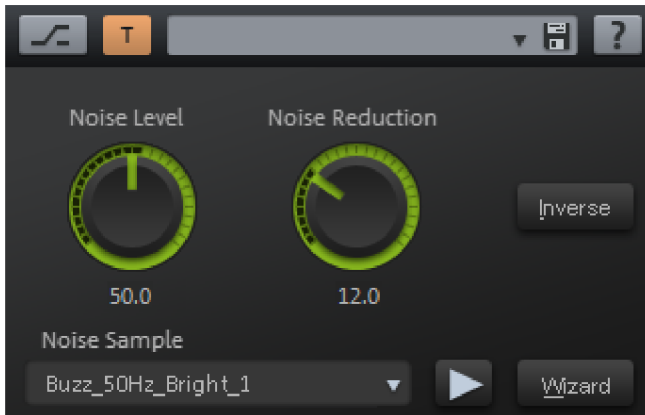
In practice, extremely quiet passages, like the slow fading out of one instrument prove to be critical – here the noise level can exceed that of the audio signal.

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

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 671)

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.

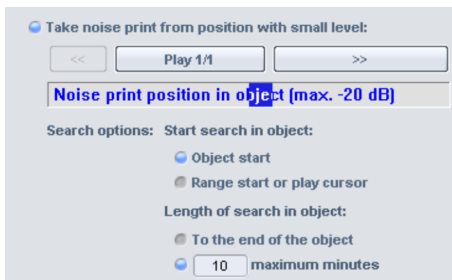
Target length: Set an explicit length for the noise sample.

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"**.

Remove DC Offset (Offline)

This function removes the DC offset in a wave project/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: Enter a minimum DC offset threshold to indicate where DC offset removal will engage. If you deselect the corresponding check, the stereo channels will be edited together. This saves computing 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

(Samplitude Pro)

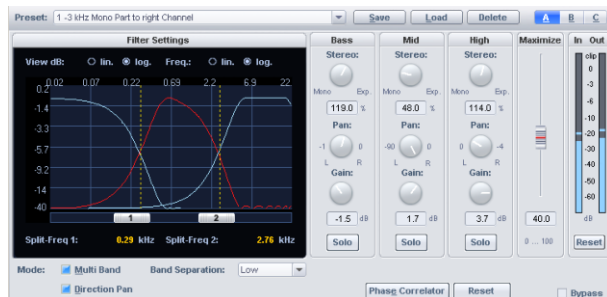
The multiband stereo enhancer performs detailed modifications and corrections to the stereo image in three independent frequency bands.

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

The preset filter settings are located in the top left of the dialog window.

The graphic shows you the frequency response curves of the individual curves. The left axis labeling shows the damping in -dB, while the upper axis labeling shows the frequencies in kHz.

The parameters for stereo manipulation and the A/B/C buttons for quick switching between the three different settings are located to the right of this.

General controls

Preset: Select from a number of presets here.

Save, load, delete: Save, load, and delete the various settings here. The preset file ending is: *.ste

A/B/C comparison: To compare different dialog configurations, switch between three different settings here.

Bass, mid, high: Set the three frequency ranges for stereo, panorama, and gain here (see below).

Maximize: Use this controller to compress the sidechain 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, frequency bands may be monitored one at a time. This simplifies the process of changing filter parameter settings. For example, 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. "Multiband" mode is not automatically active when using the enhancer in the mixer master section, which 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 included when changing the panorama setting. The panorama control functions as a directional mix controller. A centered vocal recording may thereby be moved to the left or right of the stereo balance later. The difference containing the sound sources arranged outside of the center remains unchanged.

When inactive, the panorama faders work as normal. The complete stereo signal is changed (mono and the remaining left and right portions).

Phasecorrel. (phase correlator): This opens the phase correlator. This option is very useful when "Solo" mode is active; the base width and panorama setting for each frequency band may be checked visually.

Reset: This button resets all settings.

Play/stop: Real-time preview function.

OK: The algorithm is applied to the selected range of the sample or the selected object. When using this in the master area of the mixer, the parameters of the dialog window are applied to the entire mix.

Cancel: Click this button to close the dialog window without applying the effect. When using this in the master area of the mixer, the parameters of the dialog window will not be applied to the mix.

Help: Opens the "Help" file.

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.

The multiband stereo enhancer in the master area of the mixer:

Access the stereo enhancer in the master area of the mixer by pressing the "StE" button, which is located above the master faders. Right click this button to open the stereo enhancer dialog window. The knob next to this is connected to the base width controller for the mid band.

If "Multiband" mode is not active in the dialog, the changes affect the entire signal. The knob functions as a standard base width fader.

Stereo enhancer - tips & tricks**General info concerning base width settings and standard use of the multiband stereo enhancer**

The human ear is generally incapable of perceiving 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.

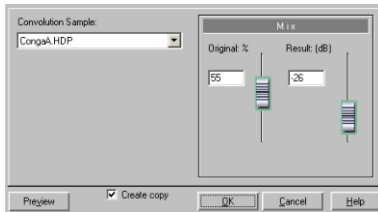
This way enables customization of samples with differing phase lengths. The function may be reversed and applied using both channels, just the left, or just the right in wave projects.

Modulation/Special

Convolution (Offline)

This menu item is hidden by default. To display it again, activate it in the menu settings via "File -> Program Preferences > Edit Keyboard Shortcuts and Menu (view page 488)".

Using the convolution function you can mix a sample/object with any other sample.



This means that the original sample will be filtered with the convolution sample: common overtones will be amplified, anything else will be softened. FFT Analysis Filter offers a similar method which filters a sample using a frequency sequence of another sample that has been saved to the clipboard. Thereby however only the frequency responses of two samples will be used. When mixed though the frequency response and the phase response are linked - which leads to noticeably different results with long samples.

With the convolution algorithm you can create reverb, echo, filter effects, exotic morphing effects and refined sonic fusions made up of different sounds. This function works along the same principles as the Room Simulator function. However, convolution calculates the exact mathematical convolution function, whereas the Room Simulator is especially optimized for reverb calculation.

Convolution sample: Here you can select the project with the convolution signal. If the convolution signal is longer than 1048576 samples, only the first 1048576 samples will be used.

Original (%). This parameter allows you to enter the level (in percent) of the original audio signal that will be mixed with the result of the convolution. A value of 0% signifies that the edited signal will contain exclusively the result of the convolution function, and a 50% value will mean that the convolution result will be mixed with the original signal at equal amplitudes.

Result (dB): This parameter sets the volume of the signal resulting from the convolution function in dB. With values exceeding -10dB, modulations can happen. For this reason, you should first select lower values.

Convolution – problems and solutions

After pressing the preview button or the OK button, the program does not seem to respond any more.

On slower computers with long impulse responses, the algorithm responds after a certain amount of time with a status display message of the progress of the calculation as it works internally on a block-by-block basis.

The output signal is overmodulated

Reduce the value of the "Results (dB)" parameter.

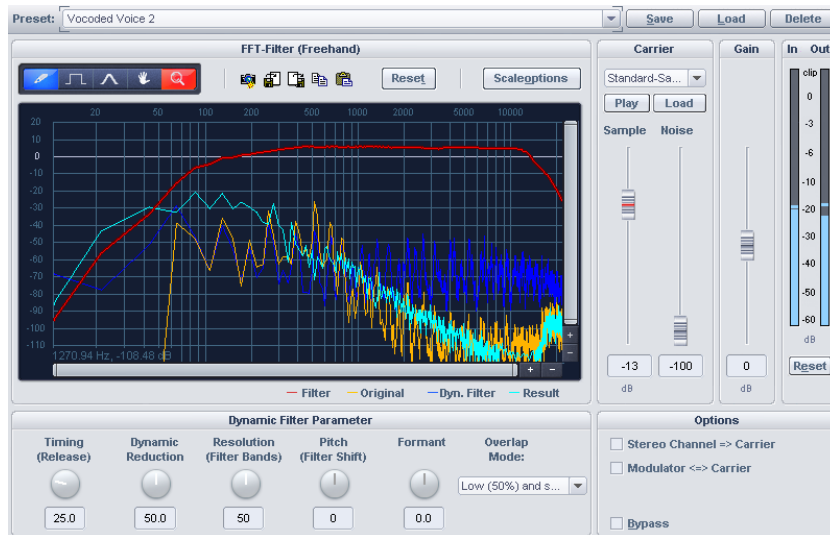
The resulting signal contains a strong DC offset

If possible, the convolution sample used should not contain any DC offset. If this is present, remove it via the corresponding "Remove DC offset (view page 672)" function.

Heavy hard disk activity or an error message concerning insufficient memory is produced

The algorithm requires ample memory, especially in case of long impulse response patterns. All background processes should be terminated and all RAM wave projects that are not needed should be closed.

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

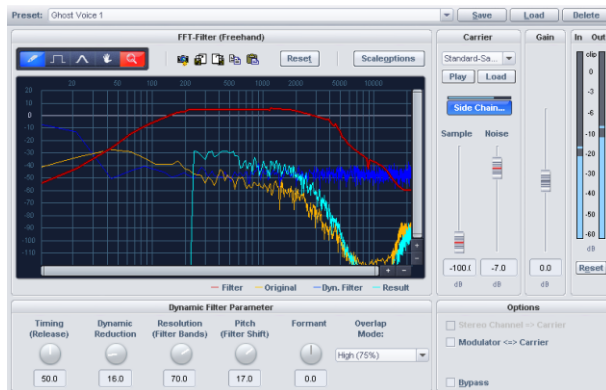
The yellow line corresponds to the frequency progression of the modulator, and the light blue line represents the carrier signal. The red line is a freely-adjustable FFT filter. The frequency progression may be drawn to optimize the vocoder results. The dark blue line is the resulting filter curve of 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.

eFX_Chorus/Flanger

Detailed information on eFX_Chorus/Flanger can be found at "[essential FX > Chorus/Flanger \(view page 685\)](#)".

eFX_Phaser

Detailed information on eFX_Phaser can be found at "[essential FX > Phaser \(view page 686\)](#)".

Corvex

More information is available in "Effects -> MAGIX plug-ins (view page 698)".

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.

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 255)".

Apply MIDI Velocity Dynamics

This command applies the current MIDI velocity dynamics settings to MIDI objects.

Set MIDI Velocity to Fixed Value

This command applies the current MIDI velocity dynamics settings next to the "Set value" button to selected MIDI objects.

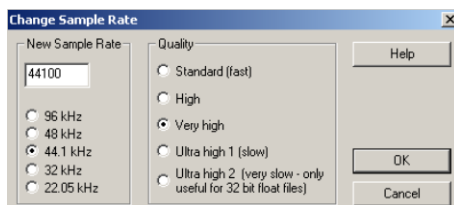
Randomize MIDI Velocity

This command applies randomly varied MIDI velocity dynamics to selected MIDI objects.

Sample manipulation

Change Sample Rate... (Offline)

Use this function to change the sample rate of a wave project.



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

If, however, you reduce the sample rate, a loss in harmonics quality occurs. When halving the sample rate of a 44.1 kHz sample to 22.05 kHz, the frequency response of the resulting signal is limited to 11.025 kHz. The frequency response always corresponds to half of the sample rate.

By clicking on "OK", you may enter the file name for the new project.

Quality

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

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

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

Very high: In this case, the linear-phase **UTR – "Ultra Transparent Resampling"** algorithm is used. This is a relatively new resampling algorithm for considerably faster and even better linear-phase, high-quality resampling. The speed increase permits use of highest quality settings even in realtime use (object resampling) that were previously only available for offline processing in predecessor versions. The quality of offline resampling also benefits from the new algorithm, since calculation in higher quality stages is now even more precise.

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

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

Note: You can make sample rate adjustments during recording and playback. The quality of the resampling used here can be set in the "File > Program Preferences > Resampling Quality Options... (view page 502)".

Sample data/2

Every second sample is removed in the wave project. Once the function has been executed, you'll see that the highly frequent components of the audio data are missing. These have fallen victim to halving the memory requirements. If you execute this command at an original sample rate of 44.1 kHz, a new sample rate of 22.05 kHz will be added. The frequency range is halved with a sample rate of 0 - 22.05 kHz changing to 0-11.025 kHz.

Sample data * 2

Use this function to insert a new sample rate between two adjacent sample rates featuring a value average of the adjacent values. The sample rate is doubled in this case. This way, the sample rate may be adapted to one another. Please note that despite the higher sample rate, no new high-frequency components come about, the frequency range of the audio data remains the same.

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.

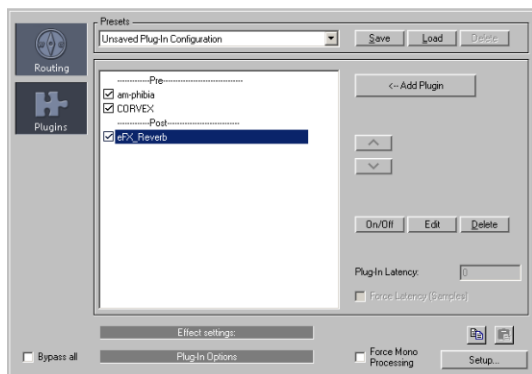
"Build Physical Loop" allows the borders of the range to be set to the precise crossover point of the sample in order to get loops without any crackling. Moreover a crossfade of the audio material at the loop end is performed with the audio material from the loop beginning, to ensure a smooth transition.

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...

Read more information about DirectX/VST plug-ins in "Mixer -> Buses and routing -> Effect routing/plugin-ins dialog -> Effects sequence/plugin-ins (view page 174)".



essentialFX

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

mode:

- **Mono chorus:** In this mode, the signal is fed through a delay unit and the pitch is modulated. Original and delay copy are mixed into a monophonic output signal.
- **Stereo chorus:** Compared to mono chorus, two copies of the original are created, modulated against each other, 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.
- **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.
- **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.

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

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.
- **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.

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

- **Input (in)/Output (out):** In this case, the input level is positioned on the plug-in or its output level. The metering precisely displays the level ratios.

Note: Input/output is available for all preset essentialFX plug-ins. The conditions are always identical to the one described here.

mode: Select the desired algorithm here. You have the choice of:

- **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.

- **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.

Stereo Delay

The stereo delay is a simple tool 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

mode: This selects between the essential algorithms.

- **Digital:** Normal, transparent delay
- **Analog:** Simulation of a bucket brigade delay (BBD). 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. But since each element of the chain led to a loss of the signal and would increase the system noise with longer delays, the devices would use a compander: At the input, the signal's dynamics are compressed, and then they are expanded again at the output end. The simulation in the eFX_delay mimics the loss and compander behavior to produce these typical audio characteristics, especially at longer delay periods and higher repetition rates (feedback).
- **delay L/delay R:** Specify the delay period for the left and right channels here.
- **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).
- **damping:** This specifies the cut-off frequency at which the highs are dampened during the delay. This 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.

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

Mode: Select the input signal to be used for detection.

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.

- **sc filter:** This fader determines the application/middle frequencies, that are filtered with the control signal for level detection. 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).

Tube Stage

Preamplifiers 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 themselves, but rather the entire circuit, which interacts with the tube in a way that amplification levels don't do on a semiconductor basis. As with guitar amplifiers, the complexity increases with the number of stages.



Optionally inside the eFX_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, 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 half 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

stages: In signal section 1 the signal only passes through the tube stages. In the second stage 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 - 192kHz.

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 like pre-eq, however, this filter is placed behind the (second) stage tube.

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 such as rough, "flat" sounds are avoided by digital gates. The eFX_Gate continuously scans the signal and automatically selects optimal settings according to the current values.

Gate parameter

sc filter Works in the same way as the eFX_Compressor.

sidechain int/ext: As with eFX_Compressor an external control input can be used for triggering.

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 the time, that 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.

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.

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

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.

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.



The eFX_DeEsser works, in contrast to other devices 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

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.

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

highpass: This fader determines the cut-off frequency for a steeply sloped highpass filter (24dB/Octaves) 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.

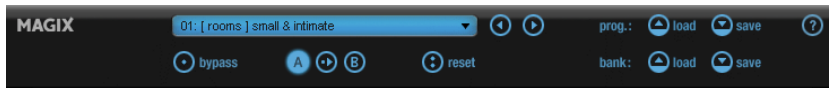
compression: For the most part, an eFX_Compressor component contributes to working parameters for optimizing voice recordings. The further the fader is turned the lower the threshold and higher the compression ratio. The Attack and Release times are selected according to the program.

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.

MAGIX Plug-Ins

Console

When opened, all MAGIX plug-ins have a global display bar at the top for managing the presets as well as for further settings, called the "Console".



A popup menu containing the current preset is located behind the display, thus enabling selection. Small up/down buttons for clicking through the presets are situated on the left.



Presets can be loaded and saved in so-called programs or banks. These terms are VST-specific. Programs are individual presets while a bank contains the entire preset memory.

For example, in accordance with the VST standard, you can change the entire memory of the plug-in by changing individual presets, replacing them with other saved ones, or saving the entire bank.



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

Further Console Elements:



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



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

Because this is the default state, "B" also contains the same settings. If you want to try out an alternative setting, simply press the "B" button: It contains the initial setting when opened. If you want to transfer the settings of "A" to "B" while testing, simply press the copy button between the two letters.



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

Parameter Smoothing

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 Pro that change delay times directly or indirectly are excluded from this action.

Vpot Controls

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

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

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

Analog Modelling Suite: AM-Track

Samplitude Pro X Suite



AM-Track analogue modeling 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

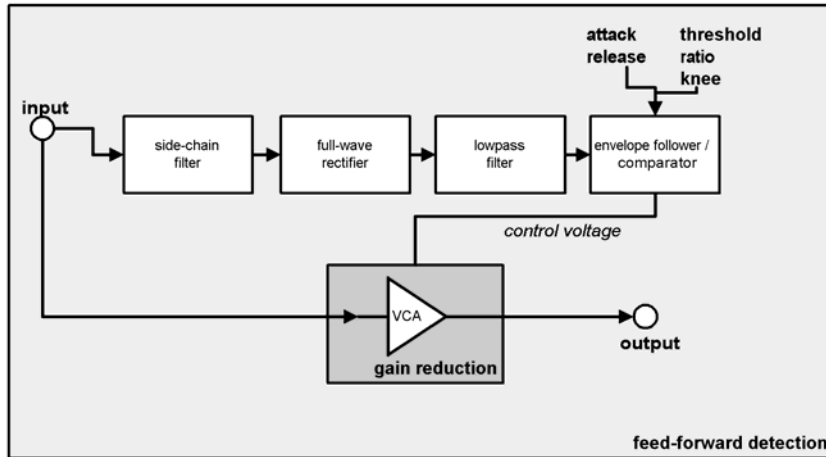


In VCA mode, the parameter selection and the circuitry design correspond to a modern compressor with a VCA element as the control circuit and a forward automatic gain control in the detector section ("feed-forward design", i.e. the controlling signal for level reduction is taken from the input signal).

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.



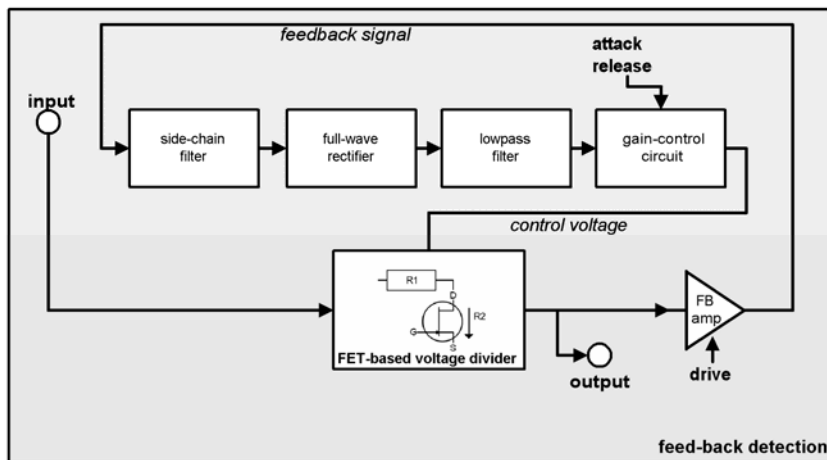
In contrast to this, there is a "feedback" method which provides a second compressor mode.

VINTAGE Mode



This mode appears as a preset when AM-Track is started. It has less 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. This, together with constant resistance at the circuit's gate, builds a so-called voltage splitter, i.e. this 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: The set time parameters for attack and release depend slightly on the program. In some cases, it's actually advantageous with 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" fader of the AM-Track regulates the so-called feedback amplification. This can be so high that the detector be 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 Ohm values. Once again, signal peaks pass through the entire circuit unaffected, but the

average level may be compressed severely. 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 less strongly compressed, 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: the output of the compressor in the signal flow features an emulation of a transformer-coupled matching-level amplifier. 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 ($10 \text{ dB} : 4 = 2.5 \text{ dB}$).
- **Attack:** This is the response time, e.g. how long the arrangement 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 arrangement 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 arrangement 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 arrangement 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 "analogue 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.

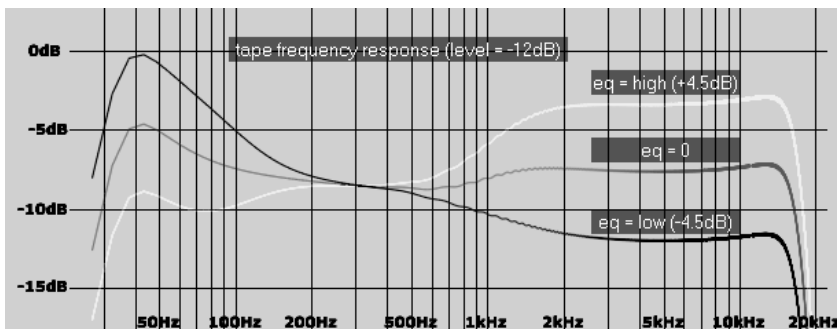
As a result of the enormous complexity of the process of magnetic recording, additional factors also influence the acoustic result, like, e.g. pre-magnetization (bias). This relates to high-frequency voltage swing (usually in sine form at 150 - 200 kHz) which is then applied to the tape via the erase 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 hysteresis 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 X Suite



AM-Pulse is a "transient modeller", 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 for sound recordings
- Sinking 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.

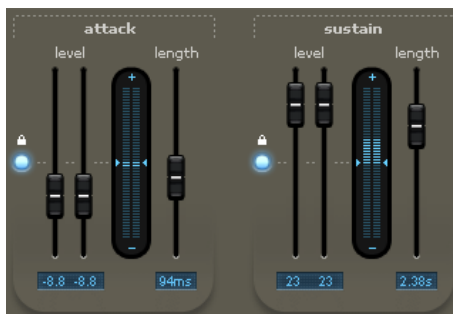
The possibilities mentioned for using AM-Pulse are based on the principle of "transient recognition".

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.

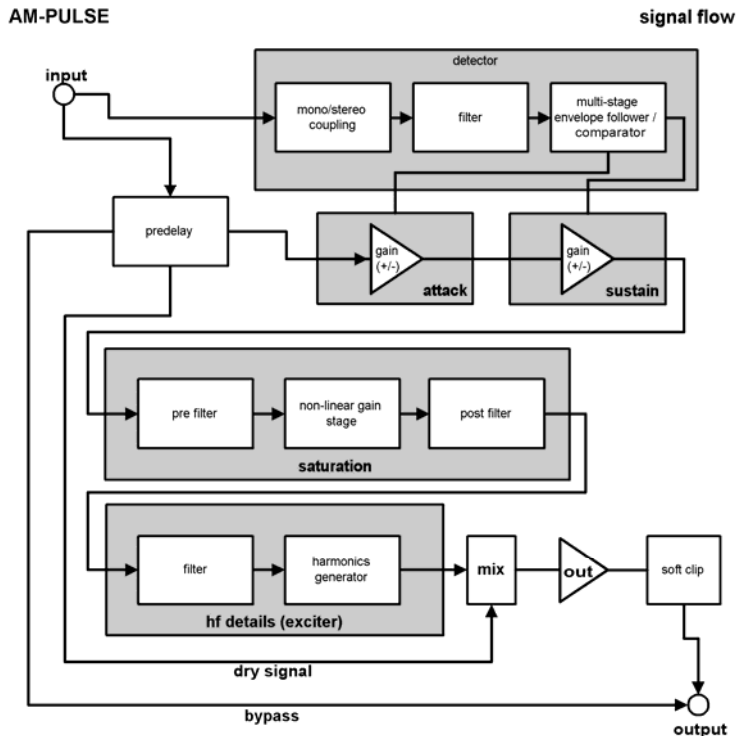
Use the **mix** fader to set how much of the processed signal is in the entire signal.

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.



Analog Modelling Suite: AM-Phibia

Samplitude Pro X Suite



AM-phibia is a tube amplifier/channel strip. It combines an optical compressor with a pre and post filter unit. By selecting the filter presets the suitable setting can be chosen depending on the input signal. While interacting with the compressor section am|phibia can be used as a vocals pre-amp, tube guitar amp or simply for creating a warm sound when mixing or mastering.

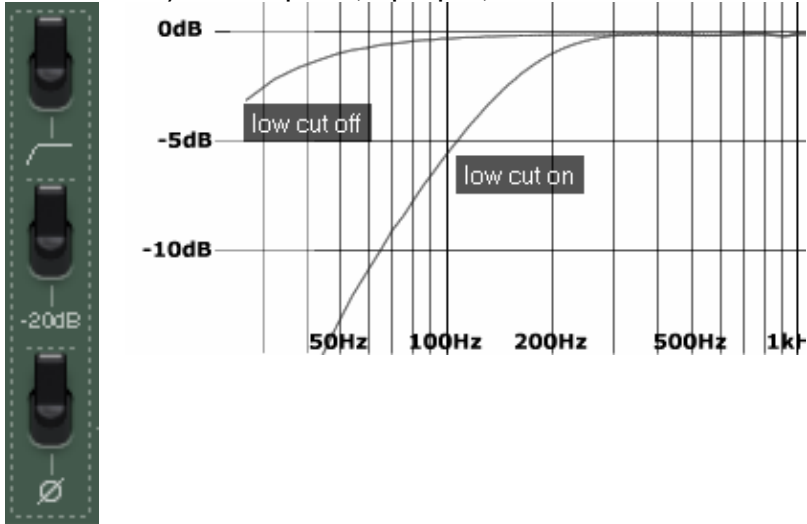
Typical examples applications of am-phibia:

- an optical compressor (the classic LDR-based light sources are modeled)
- a valve pre-amp
- a sound pre-processor with corresponding EQ and compressor settings
- valve guitar modeler with pre-amp, sound processor and case simulation
- a signal processor for warm sound synthesis with all selectable valve settings
- audio "mincer"
- Audio refiner
- Valve exciter

Many application options can be selected from a comprehensive preset list.

AM-Phibia - Signal Flow and Stages

AM-Phibia – 1) Reverse phase, input pad, low-cut



AM-Phibia – 2) optical compressor

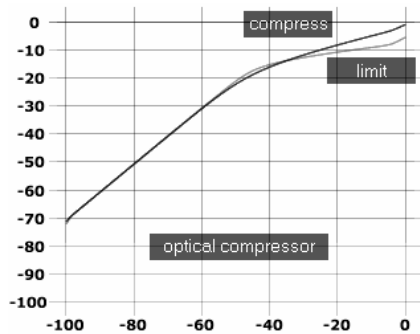


The compressor of AM-Phibia is one of the most gentle compressors available on the market and is ideal for difficult signals such as vocals. The fundamental circuit is an emulation of a classic opto-coupling design featuring a so-called feedback circuit.

For this principle a light source (LED, light bulb, or electro-luminescent panel) is supplied the output signal. The emitted light is beamed onto a light-dependent resistor and changes

to the resistance result in a change in the amplification factor of the entire circuit (controllable voltage divider). The light source is fed from the output instead of the input due to historical, practical reasons: the circuit gently oscillates to the signal and is very stable without additional settings, and operates very musically so that in hardware reality, this type of device can be built with a minimum amount of required components (which improves sound quality). Due to the overall lag of the opto-coupling system, optical compressors have a relatively sluggish release reaction. This is accompanied by the so-called memory effect of the opto-coupler. A longer exposure to light also means longer return times. Due to the feedback control, the control times and compression ratio also highly depend on the control times, and the compression ratio strongly depends on the input signal.

Specify whether you want to tap the feedback signal after the gain stage or the second filter. Frequency-dependent compression can also be achieved in combination with the pre-filter stage.



The compressor may also be switched between compression and limiter.

Note that the ratio of program regulation depends on the program.

AM-Phibia – 3) prefilter



Different types of EQ are available. Each is treated as an independently structured circuit.

- active A/B

This is an emulation of an analog filter module that operates via positive and negative feedback loops. In this case, A and B differ from one another in terms of the input frequencies of the bands. B is optimized for use with speech or singing.

- passive A/B

This circuit corresponds with the classic Baxandall network and is comparable to stereo systems and some guitar amps. Type A and type B differ slightly; the bass and highs are spaced out further spectrally for type A, and the mid range is more extensive for type B. Variant A is intended for general applications, filter B is optimized for vocals. The mid range,

which is also available in this case, is not available in a Baxandall network and has been made possible in the existing circuitry as cascading low and high-pass sections, much like in common passive equalizers. The interesting thing about this setup is the effect the individual steps have on the entire phase response, which contributes to the unique sound of this circuitry.

- guitar passive

Classic switching such as on Marshall/Fender amps. These parameters are highly dependent on each other (more highs = less bass; mids are also influenced). Just like general "passive" switches, complex phases are also present in this case, a result of variable mixing of individual branches in the filter network and the typical charm of classical design.

- guitar active

A convertible partner for a typically "American" high gain sound. The parameters do not influence each other as much as the passive variant, although a slight amount of internal feedback is applied to give the circuit a bit more power. Of course, this also influences the phase response, which also provides its own character.

- bass passive

A full-tube Peavey bass pre-amp was the source of inspiration for this EQ (T. B. Raxx, similar to Alpha), with a circuit that resembles various Marshall/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

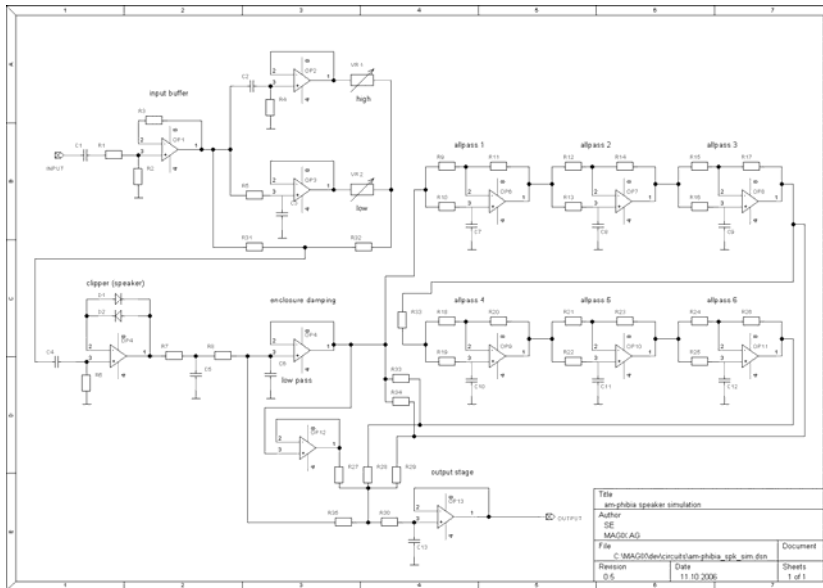
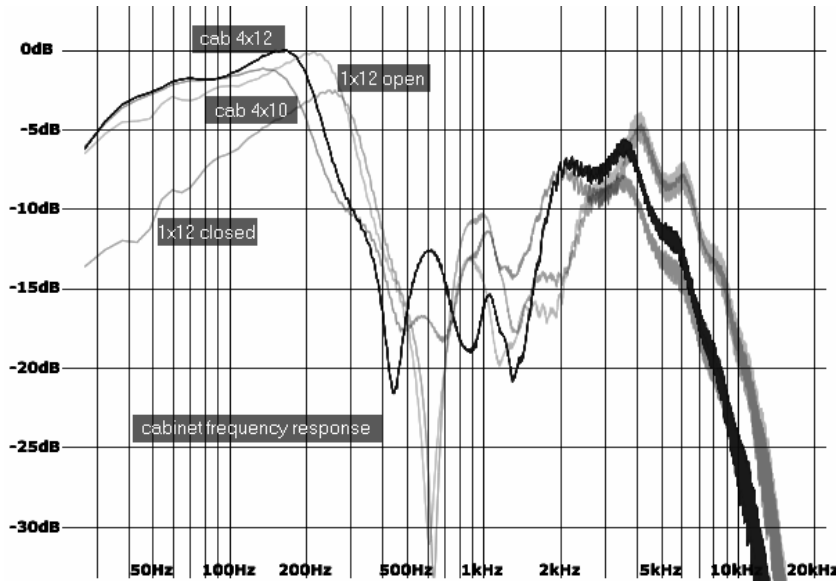
Baxandall (cut) network similar to the pre-filter, but without mids.

- 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

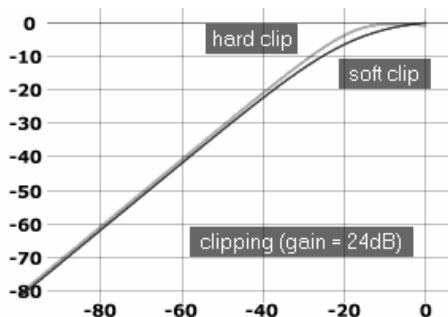
Opening this view mode gives you even more opportunities for editing. Some decisive sound parameters of the device are accessible directly from here:

- opto mem

This changes how the optical coupling device addresses the compressor and thus the related memory effect. You can use the "opto-men" 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.



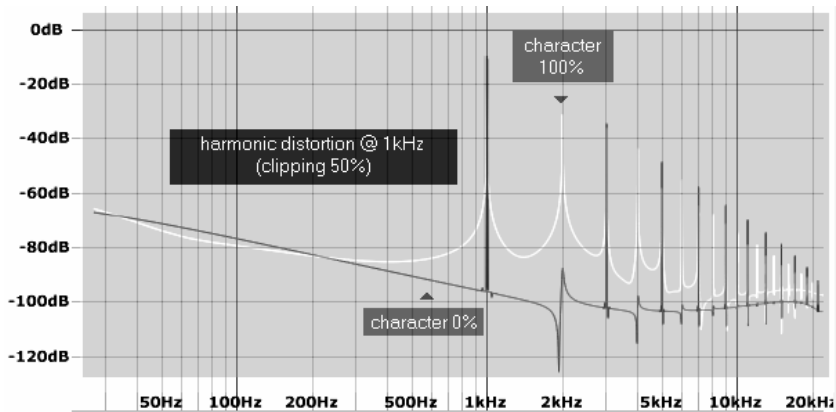
Guitar players can set this function correctly by using the volume controller of the guitar to control the distortion.

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).

Analog Modelling Suite: AM-Munition

Samplitude Pro X Suite

AM-Munition: compressor/limiter program



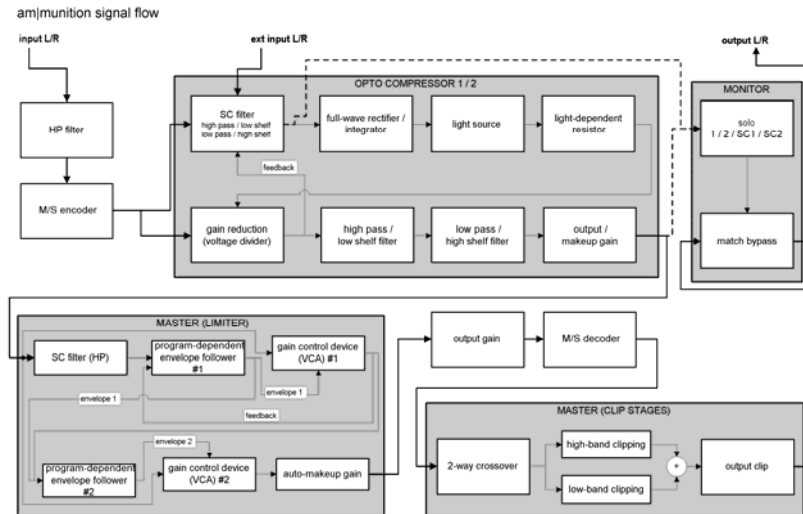
AM-Munition is a compressor/limiter for spicing up the mix and giving it more kick. This mastering and dynamics tool makes the mix not only louder and adds definition, but can also be used as a limiter. In the process, transients are conserved and a compact, but nonetheless loud signal is delivered. 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.

SIDCHAIN: 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.

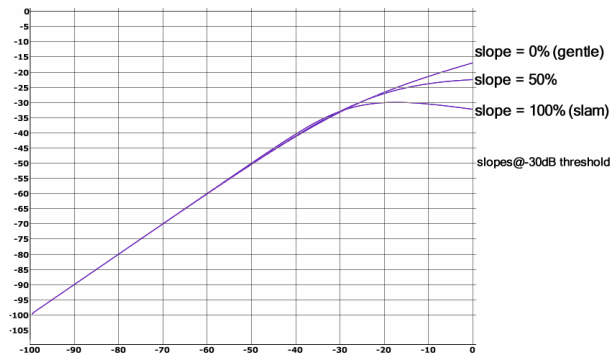
The settings of this controller affect compressor 1 and 2 and the limiter in the master section.

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". This 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

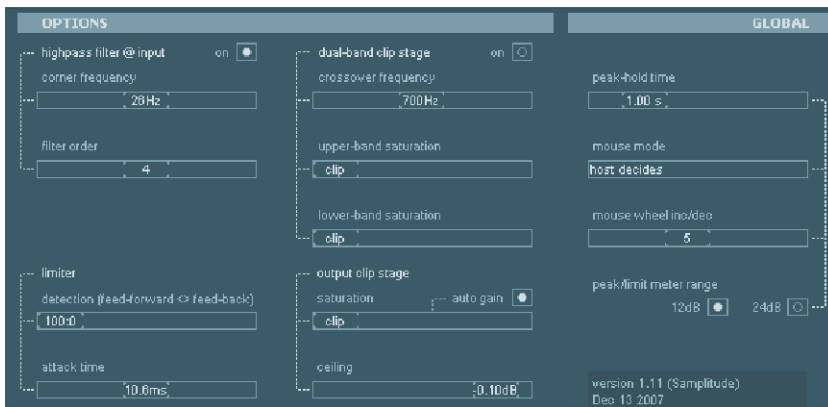
for this purpose. This works together with the corresponding controller as a signal dampener.

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 "Config" 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 settings on the right which have a global effect on the plug-in (independent of the selected preset):

Peak-hold time: The displays for output/gain reduction, clipping as well as extreme sidechain levels possess memory for top values to display the maximum values. Specify the holding period of the top value with this controller. After this time elapses, the peak hold display will fall, provided no higher value exists.

Mouse mode: Normally, AM-Munition is set according to the host's settings in terms of how the mouse pointer moves and how controls are executed. You can make these settings explicit. The list features a horizontal/vertical and circular mode.

Mouse wheel inc/dec: Use the scroll wheel of the mouse to make direct changes to wheels and sliders. Parameters allow the intensity of the value change to be determined by moving the wheel.

Peak/limit meter range: This setting determines the display range of the output peak meter, or the level reduction of the limiter (depending on the position of the "GR/Output" circuit).

The remaining area of the configuration page is reserved for **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**:

Limiters

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.

CORVEX – chorus/flanger



When referring to the chorus and the flanger effects in CORVEX, we aren't discussing only one of the two effects, but also the variations between them that go beyond the standard concept.

A chorus generates the typical "floating" sound as known from guitar sounds or synthesizer pads. You can add acoustic "depth" to an instrument to add more power to the sound or to create the illusion that multiple instances of it are present.

The chorus sound is created via the so-called Doppler effect. You've probably noticed this phenomenon in everyday life: the sound of an approaching ambulance sounds higher than when it is moving 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 equipment, this delay is within the range of 10-30 ms (as in this one), this means that it is short enough to 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 already sounds "duplicated", 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. The result is a floating effect where the speed is influenced by drifting.

The "Flange" effect is similar to that of the chorus, but does have 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 so-called 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 low delay time (less than 10 ms). "Release" or doubling of signals is not the main focus in this case; more creative deformation of the frequency response is what we are interested in.

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 Parameters

Delay/Modulation Section



Time: Specify the delay time to set whether you want a chorus or flanger sound. For the flanger, delay times are usually between 1 – 10 ms, since in this case, the typical comb filter artifacts usually lie within the audible frequency area (frequency = 1/delay in ms). For the chorus effect, values of 20 - 40 ms are normal.

It is debatable what the actual correct delay times are. However, we don't advise you to conform to any strict 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. Subsequent elements such as the filter and diffusion units also count as part of these voices/delay units. Note that an increase in the number of voices will cause increased processor load.

Span: The time of each delay unit activated per "voices" may be changed with this controller. Example: the "time" controller is used to set 10 ms and select four voices. "Span" set to 50% would mean that voice 2 would be delayed by 15 ms, voice 3 by 20 ms and voice 4 by 25 ms. . 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 may be adjusted here using the small fader below "time". You should notice one of the most important properties of these three effects here: the small faders on the lower row on the front plate features the same value. These are basically modulation targets. If the modulation target of the LFO is applied as the "time" fader, this sets the depth of the pitch shift.

At max, the amplitude is at its largest; in the minimum position, the effect remains static.

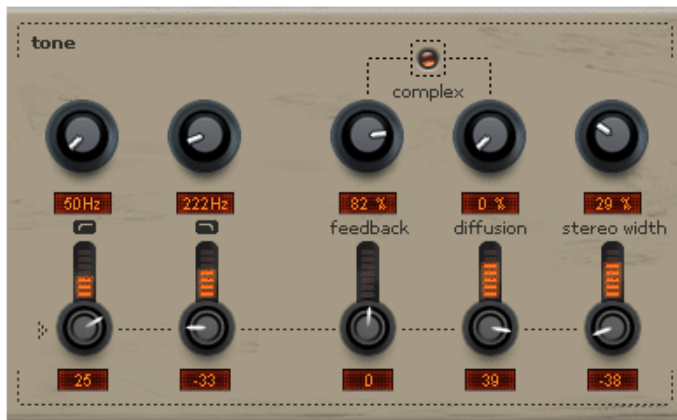


Modulation speed: The rate of modulation. Slow times create a quiet beat; high speeds sound like vibrato or, in extreme cases, like it's under water. The type of modulation may be selected using the buttons.

Waveform selection: sine, square (sqre) and random (rand) are available. The sine wave is suitable for rhythmic, quiet sounds. A square wave sounds quite drastic and very rhythmic. Random mode is useful for ambient sounds or sound landscapes, since the progressions are not foreseeable and can add an interesting accent to the sound.

The **sync** button snaps the speed to the song tempo. Alternately, you can set the modulation tempo manually in Hz.

Tone/Filter Section



The low and high-pass filters limit the signal for each voice/delay unit.

Feedback: Set how strongly the signal in each unit is fed back to itself here. High values produce a typical, cutting flanger sound. What makes CORVEX stand out from many other digital devices is that high feedback rates will not create overmodulation. For each delay unit, the signal is fed at a saturation level to create a soft, analog-sounding threshold.

Diffusion: This is one of the most interesting parameters of CORVEX, and it's rather unusual for this type of effect. 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.

Complex: Usually each voice pairing functions in "PingPong" mode, i.e. the left channel is reflected 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 eight possible voices affects every other one, i.e. echo repetitions ensure that diffusion is even faster. In this case, the maximum number of voices make drastic spatial sounds possible. For this reason, CORVEX even manages to eclipse some specialized reverb effects...

After this overview of the main parameters, we still need to explain the other small controllers on the lower row.

As mentioned above, this involves an intensity setting for 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 activation of the round LED in the modulation section results in increased delay times.
- The LFO also affects all other parameters to the right directly. To the left, the inverse value of the modulation envelope is taken. For example, if 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 modulation shift of exactly half of one period length.

Basically, the modulation values combine with the settings of the main controller. 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 opens the stereo picture to maximum every fourth note. Now turn the small controller 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 be capable of. We've created a few presets that use heavy modulation to quickly get things cooking...

ECOX – Echo/Delay

This delay provides a creative playing style that's different from common delay effects.



On playback, change the delay times 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 the tape speed to change the delay and when the system was also slightly sluggish.

ECOX can reproduce this type of sound relatively easily, and includes tracking fluctuations and loss of highs during each feedback in tape echoes, since these are essentially always present.

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 particular characteristic of ECOX is that the delay cannot be distorted "digitally". Even in a "looped" repetition, the signal cannot be distorted indefinitely, but it may be compressed by an increasingly slight degree and distorted like a tape.

ECOX - Delay Time



left + right: 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 may be selected as in CORVEX. You can also leave out "sync" and specify free delay times in milliseconds.

Link button (lock symbol): Press this button to change both channels simultaneously with a delay knob. The link function also affects the two modulation depth faders in this section.

The "sync" button beneath it affects the display of the delay time of the left and right channel in the bar grid.

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

ECOX also includes some pretty wacky presets in addition to the regular ones, just to prove how much this echo device is capable of - be creative...

FILTOX – Multimode Filter



Much like CORVEX and ECOX, FILTOX is also a "modulation effect". In this case, everything relates to frequency response deflection: a modulation source controls two filter units. 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, i.e. 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 also known as an "Oberheim" filter. With FILTOX, two of these modules were cascaded per channel to achieve a switchable 24 dB slew rate.

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.

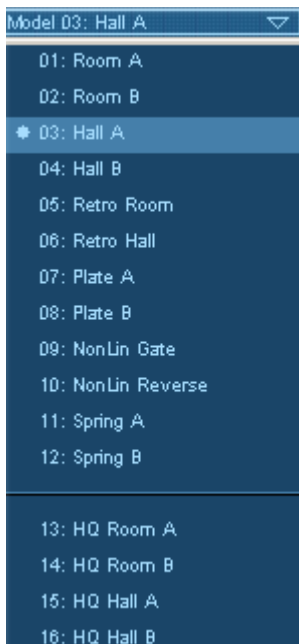
VariVerb Pro

With **VariVerb Pro** (VARiable reVERB) Samplitude offers you a virtual effect device with high quality hall algorithms to lend your recordings depth and space. Unlike the Samplitude room simulator the created reverb is not based on impulse responses but is calculated in "traditional" manner based on a complex reflection pattern and different software models.

The benefit of an algorithmic reverb such as VariVerb Pro is that you can influence different parameters and design the room impression as you like. Furthermore, many effects such as modulation or nonlinear amplification, cannot be realized with convolution, and this proves the value of algorithmic processes. The sound coloring of this conventional reverb generation in particular are an interesting aspect with regard to the aesthetic of sound.



VariVerb - Algorithms



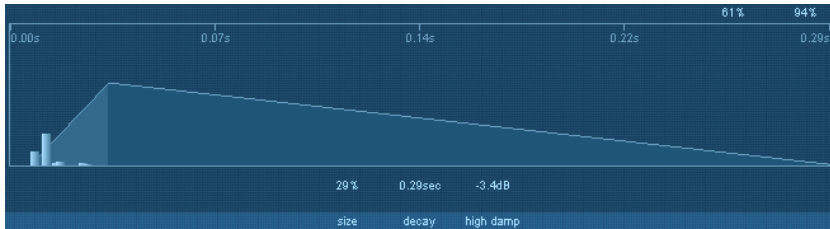
The algorithms in VariVerb Pro specialize in a specific type of room creation that may be selected from the list of "**models**" in the display:

- **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, response is slightly slower.
 - **Hall A:** Larger hall, slower reverb buildup, less density than the "rooms".
 - **Hall B:** Similar to Hall A, but features later response of the reflection, slightly "colder" sound, and less diffusion.
 - **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 similar to retro room, but a classic reverb 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, more resonance, increased vintage character.
 - **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 featuring two linked spring systems, typical echo effects and excitation noise.
 - **Spring B:** Similar to spring A, but the vibration is more balanced.
 - **HQ room A/B & hall A/B:** Extremely high-quality and very complex simulation, but subject to high processor strain. Nevertheless, or maybe because of this, it's the definitive main reverb algorithm. Very sculpted, tangible room impression featuring free positioning of the sound source and the listening location. In this case, 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.

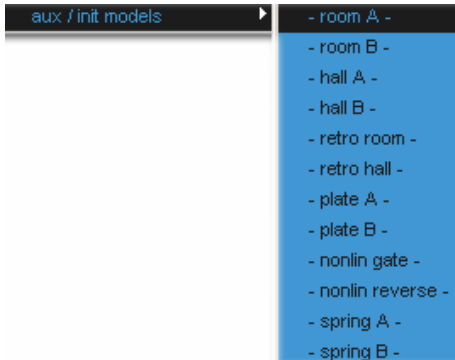
VariVerb - Overview of Details

When you open VariVerb for the first time, the device will open in a kind of "Easy" mode.



The **Display** is the core of VariVerb, since this visualizes the available parameters, their current values, and a graphical visualization of 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 **"default"** setting. This section can be used to design reverb (if you want to start from scratch). The mixing ratio is preset to 100%.



The controller for the mix ratio is located on the right of the main interface (direct signal and reverb portion).

To the left, the **"test"** section and a **"mute"** button are provided.

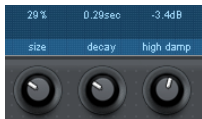


You can use these to temporarily switch off the input signal during longer reverb decay periods, i.e. if you want to judge the resonating reverb precisely.

On the other hand, a muted input can be useful for more complex room design using the test samples: these allow acoustic evaluation and fine tuning of the reverb effect using three samples: stick (sidestick), snap (finger snap), and vocal (short vocal sample).



A "Zoom" controller is located at the top left corner. You can use it to evaluate the initial reverb phase. Larger zoom steps represent the first milliseconds, and for room and hall models, these also indicate the most important initial reflections.

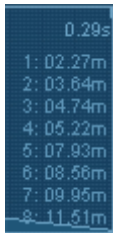


The controllers for the actual reverb parameters are located below the display. Only the most important ones are accessible first, such as room size and reverb sustain.

You can perform more drastic manipulation by clicking the **"Expert"** button above the preset list.



If you hold the "Expert" setting, VariVerb Pro presents all accessible parameters for the model it is based on.



In **"Expert"** mode, the distances for the room & hall models of the first eight reflections are displayed. The set **"size"** and pre-delay are already included. These are combined left-right values for a better overview.

Individual parameters of a model are displayed as a "tooltip" if you hover the mouse pointer over a controller. You can deactivate these tips with the same button in the display, globally if desired.



This state will be saved and is independent of the preset or the current audio project.

The following concentrates on the available algorithmic models and focuses on the 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 the simple digital process, reverberation created by VariVerb Pro is not simply produced by adding the echoes. Simple systems do not consider that real rooms compress echoes with increasing reverb duration. VariVerb Pro uses a network of reflections that very closely resemble natural processes.

The most important settings have already been set without opening "Expert" mode. To make things complete, we will list all available parameters here:

- **EQ low/high:** Allows you to "pre-filter" the signal before adding reverb. The filters have shelving characteristics. The trigger frequencies vary according to the model type.
- **Predelay:** Determines the delay time for the actual reverb process. 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 left 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.
- **Diffusion:** With this parameter, you can simulate diffusion on irregular walls and objects. The higher the value, the softer and fuller the reverb sounds.
- **ER absorption:** VariVerb calculates 16 initial reflections (compiled to 8 in the display). This controller lets you specify how broadly the last ones 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 indicates a smaller amplitude as well as damping of highs.
- **ER tail:** The mixing ratio for the early reflections and subsequent reverberation. These reverberations are decisive for perception of 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".
- **Decay:** The reverberation time. Use this controller to define how fast the reverb tail will be absorbed, i.e. the time for the reverb to ring out. Turning this knob to the left minimizes the time, you might only hear the first reflections. Turning the knob to the right minimizes damping and results in long, sustained reverberation.
- **High freq:** The threshold where the "high damp" effect starts.
- **High damp:** 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.
- **Width:** Specifies the width of the stereo effect. If set to left, the reverb is monaural.

Plate

A real plate reverb consists of a large metal plate (often 0.5 to 1 m² or more), that is activated 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 produce 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 today and have been almost entirely replaced by software simulation. The sound continues to be very popular due to the very dense sound (high diffusion) and inaudible discreet echoes. They are therefore ideal for percussive material. In case of vocals, a plate reverb generates a smooth, warm effect. The slightly "metallic" resonance of a plate can also be used to generate a vintage effect.

Parameter:

- **EQ low/high:** See room/hall
- **Pre-delay:** See room/hall
- **Size:** Size of the virtual reverb plate
- **Buildup:** Low values result in a fast reverb increase. If the value is greater, the activation phase also increases.
- **Decay:** Reverberation time, see room/hall
- **Damp freq/damping:** See room/hall
- **Width:** See room/hall
- **Modulation:** Slightly alters the pitch by automatically varying the internal delay times, comparable to a chorus effect. This can reduce possible resonance; however, depending on the material (e.g. vocals or piano) it can have a negative effect on pitch modulation.

Retro

As already indicated above, VariVerb Pro also features vintage-like algorithms in addition to realistic synthesis, which is used by many well-known manufacturers until today. 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.

NonLin

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:

- **EQ low/high:** See room/hall
- **Pre-delay:** See room/hall
- **Size:** The size of the individual "bursts" i.e. the length of the total reverb in the sum.
- **Diffusion:** Reverb smoothing effect. The greater the value, the more echo blocks are blurred and the more reverb occurs.
- **Width:** See room/hall

Please note that no decay parameter exists for these models, i.e. there is no "decay".

Spring

You may be familiar with reverb spirals from guitar and keyboard amplifiers. At the bottom of these amps, a unit consisting of two to four spirals is mounted on a vibration-free carriage. Like the reverb plate, this 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 quite peculiar sound: the typical "bloing" sound when the springs are moved, similar to splashing. When the reverb dies away, the basic pitch of the spring(s) can usually be heard quite clearly. Furthermore, the frequency range is limited due to the losses in the spirals and used pick-up/transmitter. Nevertheless, or maybe for this reason, the sound is special. Some music styles such as dub & reggae would hardly be possible without reverb.

The spring reverb is implemented in VariVerb Pro 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.

Parameter:

- **EQ low/high:** See room/hall
- **Pre-delay:** See room/hall
- **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 spread the typical "oscillation" of the spring. The length of echoes is increased.
- **Decay:** Decay length. Please note that unlike the room/reverb and plate models, an echo effect remains if the decay times are longer. This corresponds with the natural characteristic of a spring, since these continue move during the release phase.
- **Saturation:** The spring models also include the processes when "outputting" the signal and "receiving" it via the magnet/coil system. The transfer of electrical energy into mechanical power and vice versa is non-linear so that (harmonic) distortions may occur if the input level is quite 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.

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 less important, take a look at the HQ modules. These simulate a room with rare plasticity and naturalness, almost as if impulse responses were the basis. But they are not; VariVerb Pro features an armada of delays and filter stages that 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, a 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. In "Expert" view, the usual section with the reverb sustain moves to the right and makes room for viewing from above.



This includes freely positionable handles for sound sources ("L" and "R") as well as two microphones ("L mic" and "R mic"). For instance, if you increase the distance between the source and the microphone ("near/far"), you will hear the signal move away from you; as the delays increase, the sound characteristics become more "diffused" and more complex in total. 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.

Parameters of the HQ room/reverb models: Generally identical to those of regular rooms/halls.

Exception: "ER:absoptn" and "ER:tail" are not available (see explanation above).

Also available are:

- **Low damp:** Dampening level of the bass range.
- **Low freq:** Start frequency of bass dampening.

MIDI cc setup:**midi cc setup**

The MIDI CC setup is a relatively simple routing matrix, whereby 8 of the available freely configurable MIDI controllers can be used to control 8 internal parameters.



Every "Controller" field corresponds with a "Destination" field. Next, select a MIDI controller directly from one of the lists or use the learn function.

If you activate one of the buttons to learn it, any following external controller movements will be used to get the MIDI CC number. This enables one source destination pair to be learned after the other.

save setup

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 X Suite/Vandal SE in Samplitude Pro X)

Virtual guitar & bass amplifier

Vandal 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 Vandal 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.
- No use of impulse responses during speaker simulation (usually the case), but instead genuine physical modeling that calculates components such as speaker, cabinet, recording space, and microphone based on individual physical situations, since this offers enormous freedom of sound design and real-time control.
- Easy-to-learn interface combined with efficient, easy to understand 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 processing. Vandal does not create any signal delay (primarily by omitting impulse responses). Only the audio interface and its driver may be responsible for any in and out-going latency during live playing.
- Control over all component parameters via MIDI
- 4-times oversampling (below 44/48 kHz) for components that create distortion. This avoids typically digital artifacts (aliasing)

- Comparably low CPU load. Playing live requires the highest-possible drop-out prevention and as much "head room" as possible. 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 Vandal 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 770) 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 771) 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 773).

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 tides 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 stompboxes. Properties such as loss of highs in case of long delays or the "componder" 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.



TwångBång Spring Reverb: This is truly an authentic recreation of a spring-reverb system, incorporating all those lovely boingy, springy & slinky noises admired for decades. TwångBång lets you select from a spring system using two separate 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 multi-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 separate 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 necessarily 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.

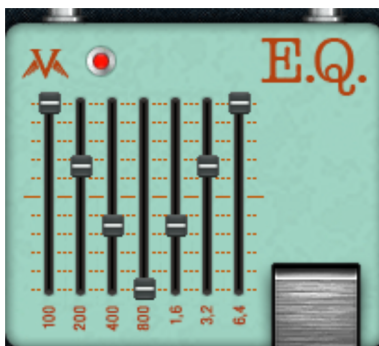
In order to keep things well-balanced and controllable, a (symmetric) push-pull circuit is used in this design.

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 may 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.

Amplifier

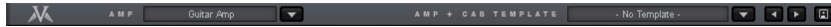
VANDAL essentially offers 2 different amplifiers; a guitar amp and a bass amp. These may be selected from the list in the "amplifier panel" below the stomp view.



As we mentioned, in VANDAL we found our own way and avoided modeling specific amplifier brands and models. The amplifiers are setup variably, allowing you to get a number of different sound characteristics out of your VANDAL amp. Internally, these circuit designs work exactly the same as the real versions. We improvised at some musically decisive points by just relying on our ears.

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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

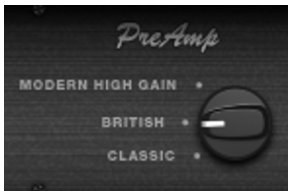


Circuit Philosophy

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.



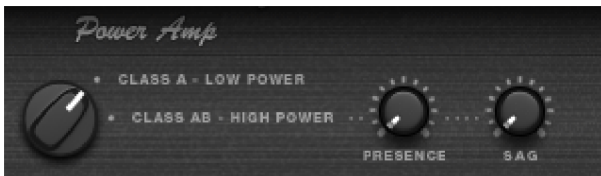
The pre-amp has three modes of operation:



- 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 761).

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.

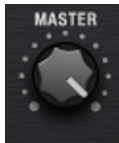
Reverb



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

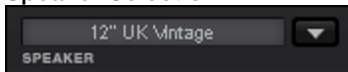


As was mentioned at the start, we went our own way in simulating the speakers and cabinets. In place of impulse responses from microphoned cabinets, we've calculated the individual components of this sort system:

- 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



The speaker selection list allows you to choose from a selection of speakers of various sizes and sound characteristics. These are organized according to guitar and bass types:

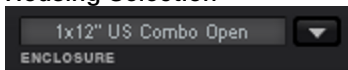
Guitar

- 10" UK Vintage 12" UK Vintage 12" UK Modern 12" UK Blue 10" US Alnico 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.

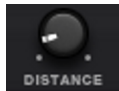
For each of the two available microphones, you may select from a list which includes several popular studio types featuring characteristic sound models:

- 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



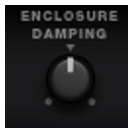
Clicking the wrench symbol accesses additional cabinet simulation parameters:



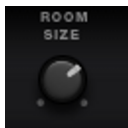
Speaker Distortion: As discussed at the start of this manual, we have also modeled the non-linear aspects of a speaker. These dramatically change the sound, depending on how hard it is working. Under heavy loads, not only more distortion is created, but the dynamics and the frequency spectrum also change. 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.



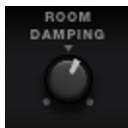
Cabinet Balance: We have already explained that the housing is an important part of the overall sound. This parameter allows you to set just how much effect this has. Turned all the way to the left, the speaker exists almost "free in space", turned to the right, you are picking up sound directly "in 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 synced 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.

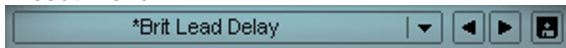
Console



In the quick start (view page 747) 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 be 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.

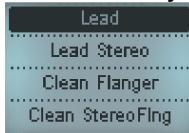
The number of possible presets available within VANDAL is unlimited. The same applies to variations to settings saved as new files. For the sake of preserving a clear overview, we recommend saving your own preset files in specific subfolders. The depth of the folder tree is unlimited.

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 771).

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 773).

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 accessing 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.

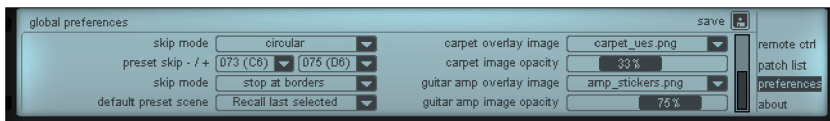
Using the "edit" menu, the current page to edit is brought up. Alternatively, an external MIDI or VST program can be sent; the underlying page is automatically picked and the selected program is highlighted in the display. Presets can be simply assigned to programs by choosing an existing file from the 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 separate 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

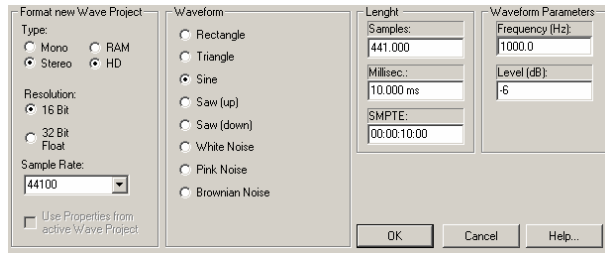


On this page, numerous in-depth settings to the software can be accessed. These act as "global" modifiers, which means they affect all (following) instances of Vandal, and they work regardless of the current chosen preset.

- startup preset: Here, the default preset is specified, which will be loaded at plugin instantiation. 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.

Waveform Generator

This dialog contains a powerful generator for test tones.



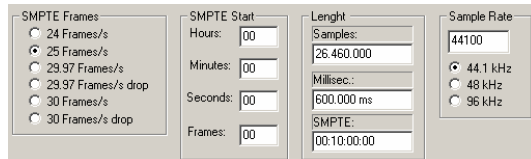
Select RAM or HD as the **type** in mono or stereo at a **resolution** of 16-bit or 32-bit float. The following **sample rates** are available: 22050, 32000, 44100, 48000, 88200, 96000, 176400, 192000, and 384000.

As a **waveform**, you choose between rectangle, triangle, sine, sawtooth (upwards), sawtooth (downwards), white noise, pink noise, and brownian 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 signal created may then be used as a wave project/virtual project.

SMPTE Generator



If you don't have an external MTC to SMPTE converter for synchronization with analog equipment, this dialog allows you to create audio files containing SMPTE audio signals. Place the generated SMPTE signal onto an empty track and route the output to the SMPTE synch input of the analog recording device.

You can choose from the following specifications as **SMPTE frames** per second: 24, 25, 29.97, 29.97 drop frame, 30, 30 drop frame.

The **SMPTE start** value may 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**: 44.1 kHz, 48 kHz, 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 792)").

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 781)" beforehand to convert this multi-track 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 to eliminate loss in sound quality. During import, the CD tracks are created as wave files and integrated into the virtual project.

If you want to import audio CD tracks into a virtual project, follow these steps:

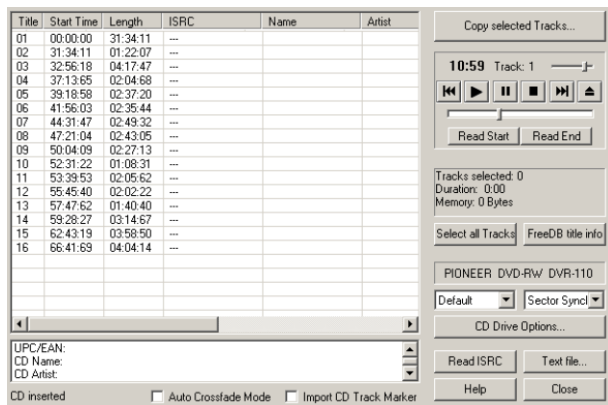
1. Open the "CD/DVD" menu or via the "File -> Load/import -> Import audio CD track(s) (view page 448)" dialog.
2. If you have more than one drive installed, click the button "CD drive options" to open the drive list dialog. Select the desired CD-ROM drive and then close the dialog by clicking on OK.

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 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 cursor keys.
4. Click on "Copy selected CD tracks..."
5. Select a file name for the created wave file or HD wave project and click 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 list and the drive list. New objects will appear in your VIP containing the audio material of you CD.

The Track List Dialog



Copy selected CD tracks: This button begins the audio copying process. All selected tracks are copied into a wave 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.

Read start/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: Click this button to access the freeDB database, which displays title information regarding the selected track.

CD drive options (view page 779): Configure the system's drive list here.

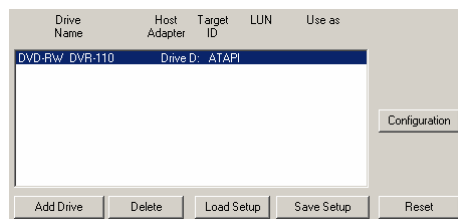
Read ISRC: This option reads the ISRC (International Standard Recording Code) for the inserted CD. This is a 12-digit ID number that provides specific information like the originating country of the label, the label's company number, the year, and a sequential title number. The ISRC code is entered into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

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

Auto-crossfade mode: Activates "Auto-crossfade" mode when importing audio tracks. This adds crossfades automatically between the tracks.

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

CD Drive Options



Configuration: This button opens the configuration dialog (view page 779) for special settings. Specify the drive name, host adapter number, SCSI ID, SCSI LUN, alias, copy mode, sectors per cycle, and sync sectors.

Reset: Restores the default settings for the drive.

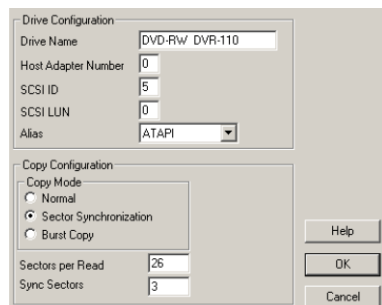
Add drive: Creates a new drive in the list that requires special settings.

Delete drive: Deletes a selected drive from the list.

Load setup: Loads the current drive list and all configuration data from a *.cfg file.

Save setup: Saves the current drive list and all configuration data in a *.cfg file.

CD-ROM Configuration Dialog



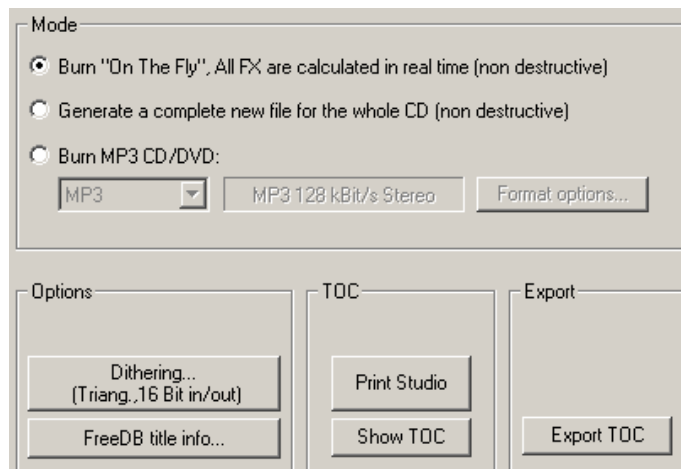
1. Open the dialog via "CD/DVD" or via "File -> Load/import -> Import audio DVD (view page 448)".
2. If more than one drive is installed, click the "DVD drive" list to open the drive list dialog. Select the desired DVD drive.
3. Select the desired track(s) using the keyboard combination "Shift" or "Alt" 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 498)".

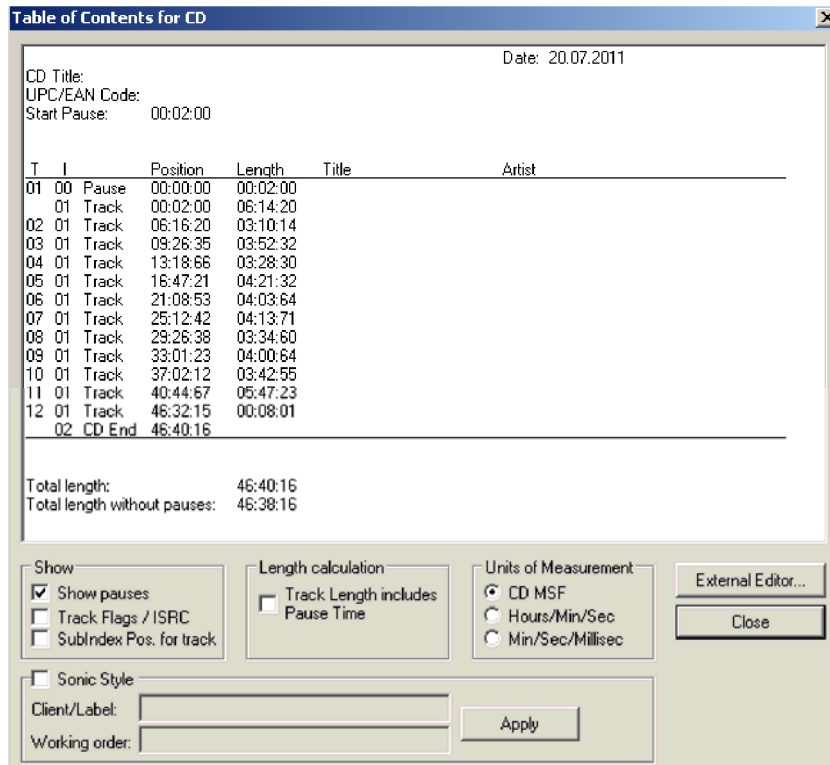
freeDB title info: This function causes Samplitude to search the freeDB.org database for information that corresponds to the track and displays it if found.

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 778)
- Sub-index position in the Track

Under "Length calculation" you can select the "CD track length includes pauses" checkbox. This means that the time span between CD pause index and CD track index is factored into the length of the track. The following measurement units are available: "CD MSF", "Hours/Min/Sec" and "Min/Sec/MilliseC".

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

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

TOC - Export

Pressing the "TOC export" button saves the information on the CD's contents as a *.toc file.

Burn CD

Recording Options

Samplitude is going to record a CD-R according to your specifications.
Please select appropriate settings for recording speed and simulation mode and press the 'Write' button to continue.

Recording speed

For CD burning "On The Fly" CPU usage while playback should be below the following values:

1X (75 frames/sec) 90%

2X (150 frames/sec) 45%

4X (300 frames/sec) 20%

Max: 40 x 10%

Recording options

Simulate recording Test and compare CD after burning

Prepare CD-R for additional data session(s) (e.g. for CD Extra)

Activate Burn Proof Burn with CD Text

Number of copies: 1

CD Text Settings

Write Cancel

CDR write settings -> Simulate write process: Use this option to test the various write speeds.

Test and compare CD after burning: This writes a temporary wave file to the hard disk while burning a CD.

Note: This function requires increased performance when burning CDs, especially at higher burn speeds. The burned CD is then exported digitally and compared with the temporary file which is then deleted.

Prepare CDR for subsequent data sessions: 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 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 792) editor.

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

Create DVD-Audio...

Samplitude enables you to burn DVD audio discs with any DVD burner. +R/-R/+RW/-RW formats may be processed.

Note: DVD audio discs can only be played on players with DVD audio capabilities! These players can be identified by a visible DVD audio logo somewhere on the device.

For Surround sound playback your DVD audio player has to have several 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".

Audio may be recorded on a DVD audio disc in 16 and 24-bit rates at sample rates of 44.1 and 48 kHz, and double and quadruple these rates are also possible. Up to 6 discrete audio channels are supported in 5.1 Surround, stereo, or 4.0.

The only limitation is that the maximum guaranteed data rate of approx. 10 Mbit/s for hardware players may not 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.

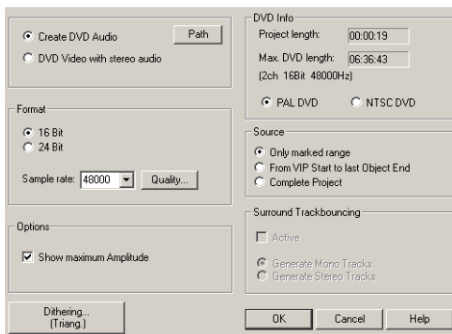
When creating an audio DVD, the "trackbouncing" dialog will open. Select either "**Create DVD audio**" or "**DVD video with stereo PCM**".

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.

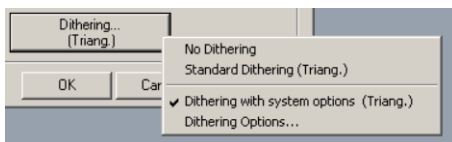
An external program will open if "**Path**" is pressed. The path for this program is set but can be changed.



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 498)".

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 track index

Use this function to set a track marker at the position of the play cursor (index markers). 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 789)" 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 790)".

Shortcut: Ctrl + Alt + I

Set CD Sub Index

Choose this option to place a sub-index marker at the current play cursor 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 play cursor position.

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

Set CD End Index

This 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

T: 1	001:01:000
T: 2	003:03:000
T: 3	010:01:000
T: 4	013:02:000

013:02:000	<small>Start</small>	<small>Book</small>	
------------	----------------------	---------------------	--

CD Index Options	
<input type="checkbox"/> Preemphasis	<input type="checkbox"/> Copy Protection
<input type="checkbox"/> ISRC Code	<input type="checkbox"/> Sec. Gen. Prot.
	CD Text
	Apply to all

Create			
Marker	CD Track	CD SubIndex	CD Pause

Move Track	<<	>>
------------	----	----

Delete selected	Play Marker	Stop
Delete all	Play Loop	OK

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 778)**: These settings can be transferred to all indices.

Pre-emphasis means that high frequencies are accentuated before 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.

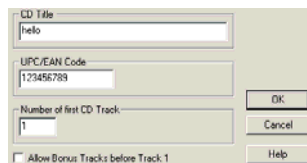
"**Play loop**" plays a loop around the marker.

"**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)

The screenshot shows a dialog box with two main sections. The top section is titled 'Selection:' and contains a list of tracks: Track 01 - 001:01:000, Track 02 - 017:01:176, Track 03 - 026:02:229, Track 04 - 037:04:022, Track 05 - 045:02:275, and Track 06 - 054:02:207. To the right of this list are buttons for 'Previous Track', 'Next Track', 'Copy Data to all Tracks', 'OK', 'Cancel', 'Load...', and 'Save...'. The bottom section is titled 'Disc / Album' and contains several input fields: 'Title:' (hello), 'Artist:' (thmïü), 'Song Writer:', 'Composer:' (pug), 'Arranger:' (opä), 'Messages:', 'Genre: (CD Text):' (Alternative Rock), 'Genre: (MP3 ID):' (Alternative), 'Closed/Year:' (2007), 'Disc ID:' (001), and 'Supplemental Genre:'.

Note: The number of maximum available characters for the entire CD is 2824.

If you can use 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.

Get freeDB Title Info

Use this command to start a search request where the freeDB database searches for and provides information on tracks from you project.

freeDB acquires information on CD track names and artists.

- In the CD import dialog, you can request information about the desired CD.
- In the CD burning dialog and in the CD/DVD menu, you can search based on the set track markers.
- An inverse search is also possible with "**Online CD search**" in the arranger. You can set track markers automatically for known albums.

freeDB Options

Submit CD to freeDB

To send CD informations please follow these steps:
 1. Enter and check all information below
 2. Press submit
 3. Send the automatically generated S to freedb-submit@freedb.org

CD Info

DiscID	Album name	<input type="text"/>
<input type="text" value="0d000003"/>	Artist	<input type="text"/>
Number of	Genre	<input type="text" value="3"/> <input type="text"/>
<input type="text" value="3"/>	Category	<input type="text"/>
Revision	Year	<input type="text"/>
<input type="text" value="0"/>	Comment	<input type="text"/>

Track Information

<input type="text" value="Track 1"/>	Title	<input type="text"/>
<input type="text" value="next >>"/>	Artist	<input type="text"/>
	Comment	<input type="text"/>

Enter all of the CD's information into the dialog. Press the "Submit" button to transfer the data to freeDB.

freeDB User Data

The entries for username, local computer name, and email address are used for communication with the freeDB server.

freeDB Proxy Options

Enter server data like server, port number, server list, and timeout here.

Clear freeDB Cache

With this function you can delete the cached data from freeDB.

Search CD Online and Set Track Indices

Search for a CD online by entering the CD name or artist. The track markers are requested from the online database. If the search was successful, click "Disc ID" for the desired CD. You'll now see a text page with the track data.

If you copy the link from your browser into the text field in the dialog, you'll be able to apply the CD track markers of the located CD to your arrangement.

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.

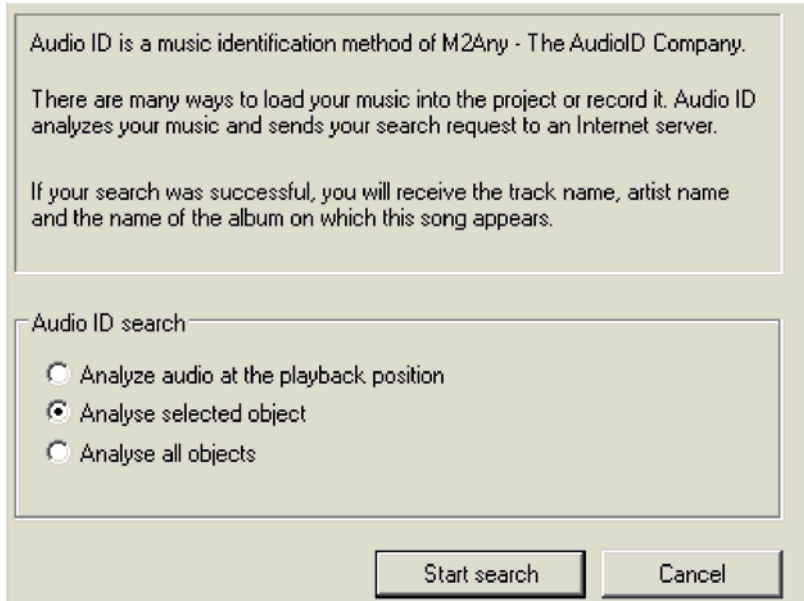
Show CDR Disc Information...

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).

audioid

audioid is part of the MPEG-7 standard and for identifying audio material. In this case, audio data is recognized automatically and associated information, e. g. track or artist name, is provided in real time.



Here are the options from which you can choose:

- Analyze audio at the playback position
- Analyze selected object
- Analyze all objects

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 158)".

Shortcut: M

Transport Console



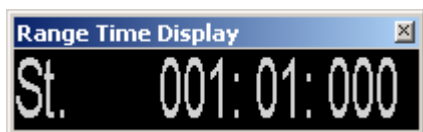
Shortcut: Ctrl + Shift + T

Detailed information about the transport control is provided in "Screen elements -> Program interface -> Overview -> Transport console (view page 52)".

Time Display

Shortcut: Ctrl + Shift + Z

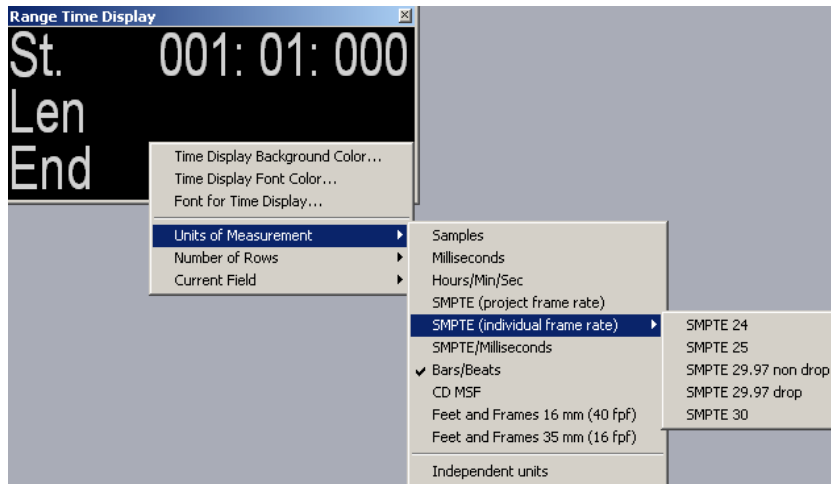
This function shows a zoom-enabled time display.



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.

Double clicking lets you select and edit most sizes.



Time display – current field

Position/start of range: Displays the current position/current play cursor or the beginning of the range. When moving the objects, you can see the starting position of the object here. Enter a negative number into the range when editing and the play cursor will be set to the end of the range.

Range length/object length: Specify the range length/object length here. If a negative amount is entered, the range end will be displayed instead of the range start.

Range end: Set the range end/object end here. If a negative amount is entered, the range start will be displayed instead of the range end.

CD position: This displays the following current positions:

- Position from CD start
- Position to CD end
- Position from current CD track
- Position to CD track end

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).

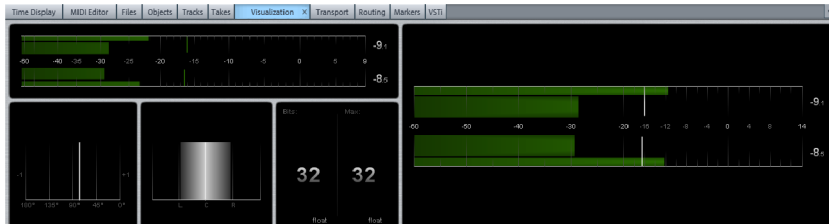
Recording position: This value displays the current recording position.

Remaining recording time: This value indicates how long recording may continue until the hard disk is full.

Visualization

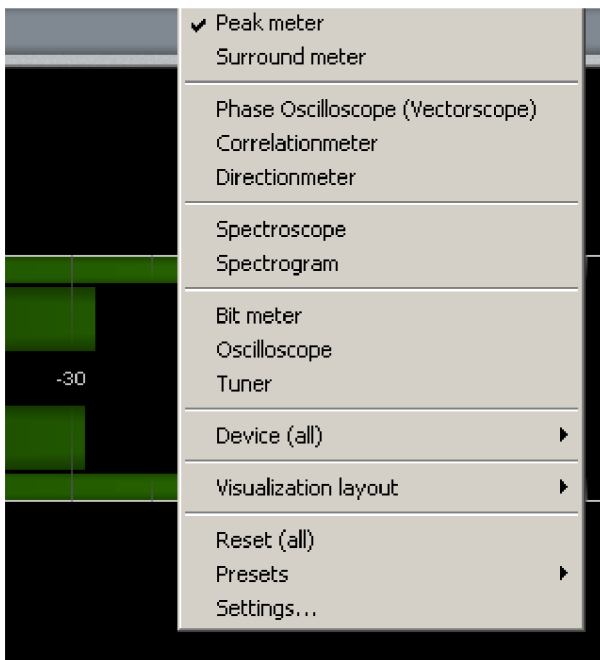
Shortcut: Ctrl + Alt + Shift + V

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 Docking Bay (view page 51) or open it in its own window.



The visualization is, as standard, sub-divided into five display windows. Choose from the following display options for every window: **Peak meter** (control display), **Loudness meter** (only Sequoia), **Oscilloscope**, **Bit meter**, **Correlation meter**, **L/R oscilloscope** (phase correlation), **Surround meter**, **Direction meter**, **Spectroscope**, **Spectrogram** and **Tuner**.

The visualization display can be adapted easily to your personal presets. **Right click on one of the visualization windows to open up a context menu**, in which you can change the window's settings. The size of the window can be changed by dragging its edges in the desired direction.

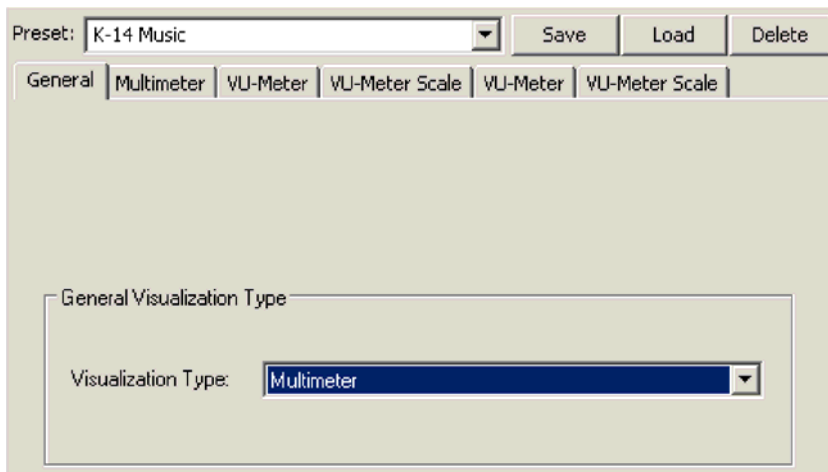


Device: Here you can set which output the visualizations should refer to. With this option you can determine which device should be displayed if multiple audio devices available. If you, for example, mix more than four submix busses that are each routed to a different device, you can display each bus individually.

Visualization layout: Here you can **load preset layouts**, **save individualized layouts** or modify layouts by **adding** or **removing columns** and **rows**.

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

Settings....: Select this menu entry to open the visualization settings dialog.



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

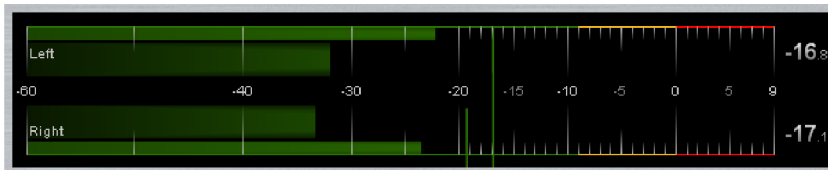
Furthermore you can load or delete the settings using the corresponding buttons.

Load/Save settings

If you have saved your own presets in the available "fx-presets" directory, these will be available here under "**Load setup / Save settings**". Various presets are also provided, for example the **K-metering settings K-12, K-14, and K-20** (view page 803).

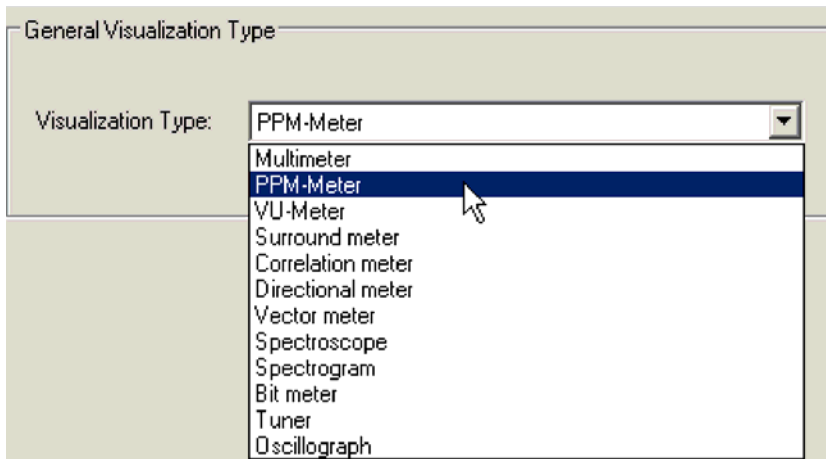
Activate docking for visualizations: With this option you can turn on docking for visualizations.

Peak meter

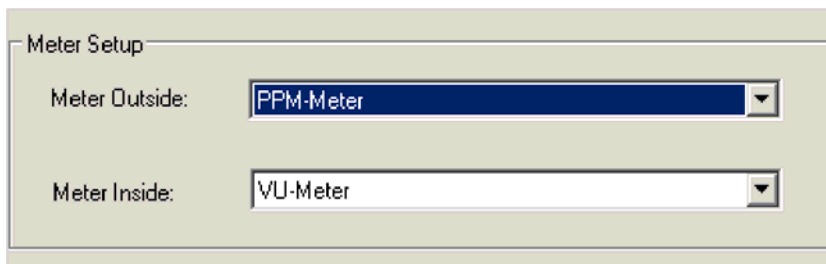


The (Multi-) peakmeter (Instrument multimeter) displays the volume during playback in dB. The thinner, outer bars show the **Program Peak Meter (PPM Meter)** whilst the thicker, inner bars display the **VU meter**, as you may still know from your analog equipment.

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

Preset: K-14 Music Save Load Delete

General PPM-Meter PPM-Meter Scale

Pre-amplification:
Headroom: 0.0 dB

Time constants:
Integration Time (Approx-PPM): 0.0 (Sample) ms
Response Time: 13.33 dB/s
Peakhold: 2500.0 ms

DC Filter:
Cut-off frequency: 0 (Off) Hz

Modulation display:
Overmodulation threshold: 0.0 (Full Scale) dB
Minimal amount of overmodulated samples: 1

True Peak metering
 Activate (Peak meter works with 4 times the oversampling)

Apply OK Cancel

Headroom (dB): With this you can initiate a level offset which will be globally added before measuring. This is the reason why different systems are calibrated differently. A value of 9.0 (IRT) e.g. ensures that the level with + 9 dB is calculated in such a way that the extra 9 dB has got more headroom at its disposal, provided your player is accordingly calibrated.

Integration time - PPM (Mms): This constant ensures that the peakmeter's 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 smaller you select this value, the slower the display bar of the peak meter move when returning. A typical value would be e. g. 13,3, which corresponds to a release time of 1,5 seconds at 20 dB.

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

Cut-off frequency (Hz): With this highpass filter you can filter out the DC component so that it doesn't enter the peakmeter measuring.

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

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

Activate True Peak Measuring: In True Peak mode you can measure the peak meter with fourfold oversampling.

PPM-Meter Scale

You can choose different display options on this page.

The image shows a software interface for configuring the PPM-Meter Scale. At the top, there are several tabs: 'General', 'Multimeter', 'PPM-Meter', 'PPM-Meter Scale' (which is selected), 'VU-Meter', and 'VU-Meter Scale'. Below the tabs, the 'Scale' section contains a 'Preset' dropdown menu set to 'K-20'. The 'Scale Range' section contains three input fields: 'Reference Level (Scale)' set to '23.00' dB, 'Minimum' set to '-60' dB, and 'Maximum' set to '20' dB.

Scale: Select from a range or different scale display settings.

Reference level (Scale offset): Enter the value for the reference level here.

Minimum/Maximum: Here you can set the min/max value. You can restrict or expand the scale's range value.

Color settings: Enter the colors and thresholds for the overmodulated, critical, optimal and undermodulated ranges.

The screenshot shows a 'Color Settings' dialog box with the following configuration:

- Overmodulation Range:** Color: Red, Threshold Level: 4.00 dB
- Critical Range:** Color: Yellow, Threshold Level: -9.00 dB
- Optimal Range:** Color: Green, Threshold Level: -inf dB
- Undermodulated Range:** Color: Blue

At the bottom, there is a checkbox labeled 'Disable enhanced bargraph design' which is currently unchecked.

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 undermodulated range, should it fall below the set threshold level.

VU-Meter

On this page, just as with the PPM meter, the parameters **Headroom**, **Integration time**, **Peak hold** DC-Filter and **Cut-off frequency** are available.

VU-Meter Scale

On this page you can, just as with the PPM-meter scale, select the display options and change the color codes for the bar graph, which employs the same scale representation as the external meter.

K-Metering

In the visualization area of the peak meter, there are a number of presets for the K-metering system. Load the supplied presets via the "Settings..." dialog of the context menu, to implement metering according to K-12, K-14 and K-20 norms. The corresponding presets can be found in the "**fx-preset > K-12.vis/K-14.vis/K-20.vis**" folder. Uniform reference volumes on different media can be achieved especially when mastering under normalized listening conditions. The peakhold 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. With K-System Metering

it's not so much the loudness, but rather the musical dynamics that take center stage. Setting the level to 0 dB sets the reference volume which no longer matches the maximum level, as often used to be the case.

According to the program and audio material, three different meter scales can be used:

K-20 = 0dB reference (83dBSPL) at -20dBFS

K-20 is especially recommended when using audio with large dynamics like classical music or film sound.

K-14 = 0dB-Reference (83dBSPL) at -14dBFS

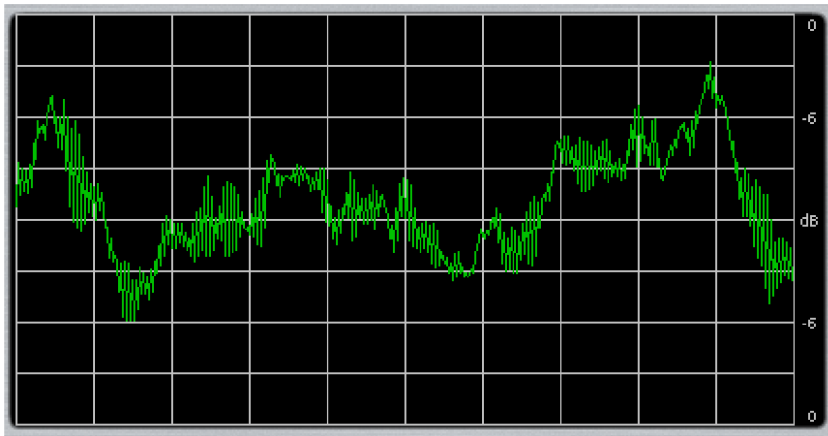
K-14 is especially recommended for rock and pop productions or for Surround Sound.

K-12 = 0 dB reference (83 dBSPL) at -12 dBFS

K-12 is especially recommended for radio and television networks.

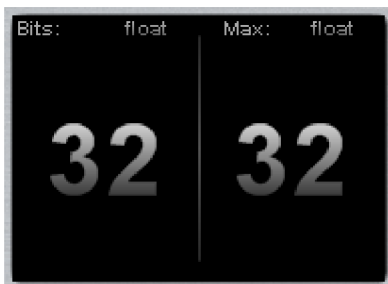
The corresponding scale for setting up the monitoring volume is calibrated with pink noise. If you set pink noise to 0dB, you get a level of 83dB SPL, a volume reference that originated in the film world.

Oscilloscope:



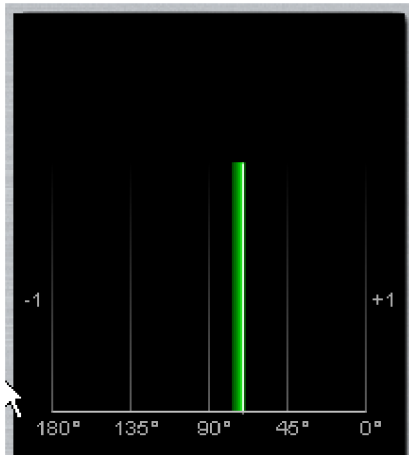
The oscilloscope displays the signal amplitude over the time curve.

Bit Meter



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

Correlation meter

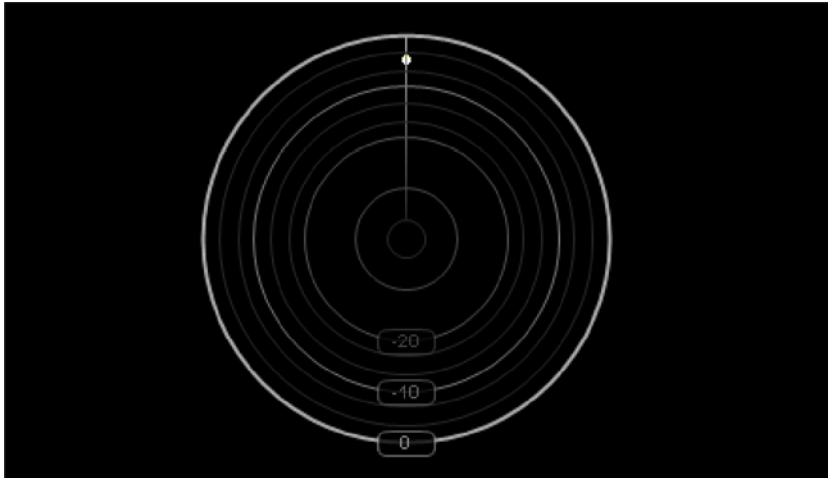


With the correlation meter you can read phase offsets between the two stereo channels, and examine how much has been deleted. A value of **+1** corresponds with 0° phase shifting. The value **-1** corresponds with 180° phase shifting. If the signal display is in the left, red area between 90° and 180° the signal will no longer be reproduced properly over a mono receiver.

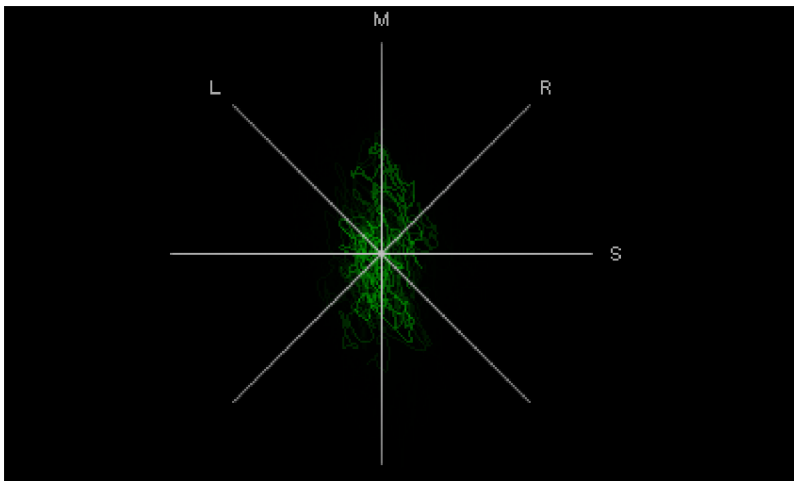
In the settings dialog you can select the colors for Mono and Stereo and mono compatible ranges, threshold values can be set too.

Surround meter

The surround meter displays the signal using the surround settings that are set as global. It shows each speaker's level in a polygon on a sound representative area.



L/R Oscilloscope (phase correlation):

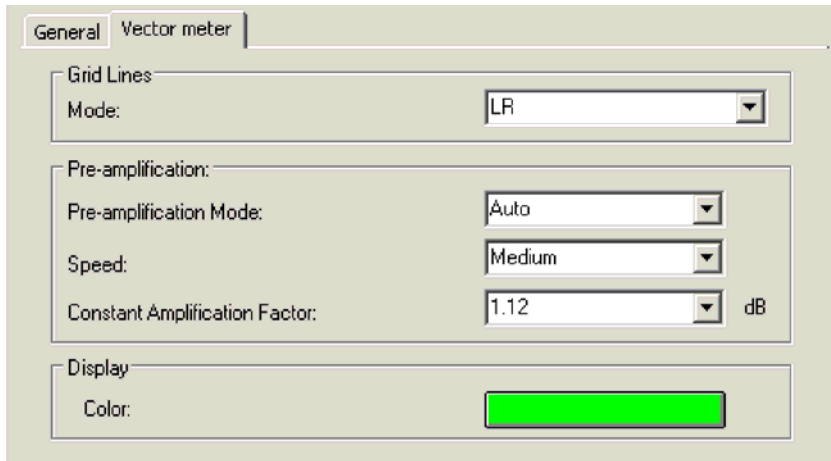


Phase correlation provides information on the distribution of the stereo image in your recording. A mono recording would be displayed as a vertical bar in this view. A song produced in stereo is by contrast shown as "diffuse ball," because multiple instruments were assigned to the mix in different panoramic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

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

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

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

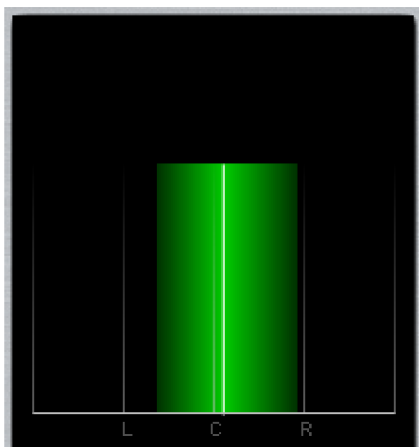


The image is correspondingly increased in pre-amplification mode, to ensure readability. You can also switch off this mode and set a constant amplification factor instead.

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

Finally you can set the display color.

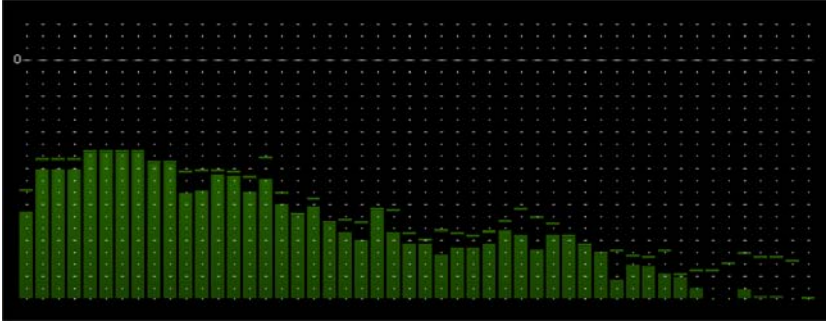
Direction meter



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

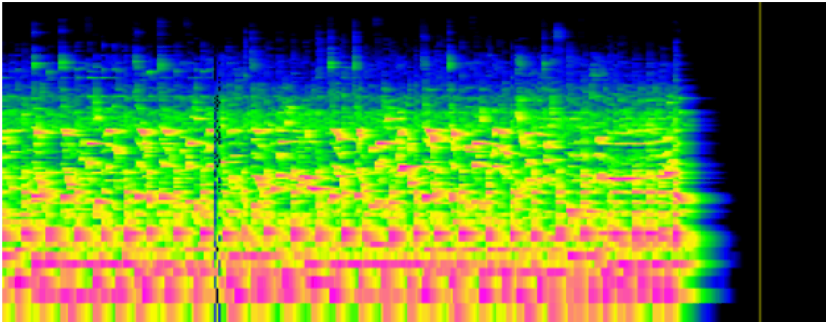
In the settings dialog you can select the colors for Mono and Stereo and mono compatible ranges, threshold values can be set too.

Spectroscope



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

Spectrogram

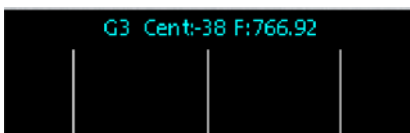


In the Spectrogram the signal is displayed as frequency proportions in a time curve. The volume of frequencies is visualized by its brightness.

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 72).

Tuner



The tuner displays the corresponding pitch levels featured by the signal. Use the visualization to tune a guitar or other instruments.

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

Manager/Docker

You can open the Manager/Docker here. In addition to the Manager windows (view page 138) 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)

Detailed information on the individual manager sub-windows is provided in the "Manager (view page 138)" chapter.

New Manager...

Use this function to open a new manager window.

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.

Positionbar

Use this command to hide or show the position bar using positioning and zoom buttons. A check after this menu point indicates that the status bar is visible.

Positionbar 2

Use this command to hide or show the second position bar and the positioning and zoom buttons. A check after this menu point indicates that the status bar is visible.

Punch/Playbar

Use this command to hide or show the punch/playback bar using the button for various playback functions and punch recordings. A tick after this menu point shows that the status bar is visible.

Mouse Mode Toolbar

Use this command to hide or show the mouse mode bar to select the individual mouse modes. A check behind the menu item indicates that the mouse mode bar is visible.

Rangebar

Use this command to hide or show the range bar for selecting ranges. A check behind the menu item indicates that the range bar is visible.

Snap bar

Use this command to hide or show the grid bar.

Workspacebar

Use this command to hide or show the workspace bar; this provides various tools for button assignment according to application. A check behind the menu item indicates that the workspace bar is visible.

Buttonbar

Use this command to hide or show the button bar featuring the track editor, object editor, MIDI editor, visualization, transport control, mixer, and manager. A check behind the menu item indicates that the button bar 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 parameters (view page 578)".

Keyboard shortcut: Scroll lock

Autoscroll Soft

Toggles between page and soft autoscrolling.

See "Playback -> Playback parameters (view page 578)".

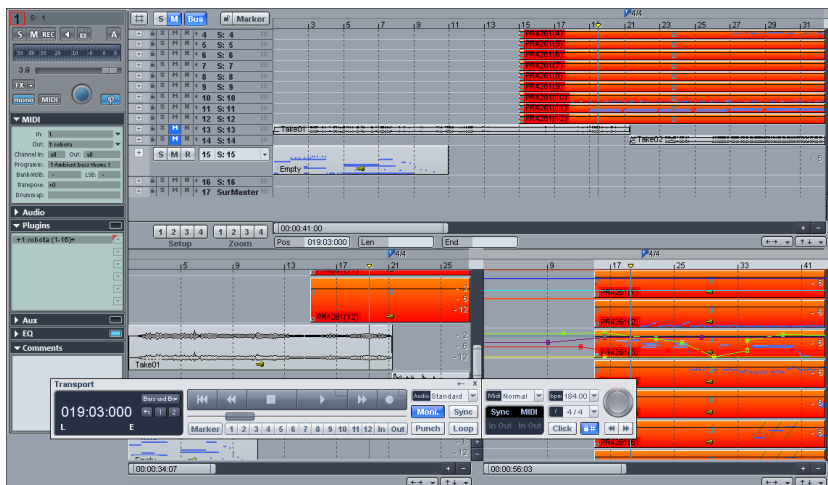
Keyboard shortcut: Shift + Scroll lock

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.

Clips

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 (clips)", 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 (clips)**" 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 be 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

The menu commands "Activate previous clip" and "Activate next clip" switch the active segment.

Activate destination

This command activates the previous/upper clip.

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 tracks.

Attention: This function keeps bus tracks hidden, even if a check is set in the track manager next to the "Arrangement" column.

Grid view

Show Grid

This command activates the coordinate grid for the project window.

Shortcut: #

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 play cursor (position line).

Snap Active

This functions turns the grid on or off.

Shortcut: Ctrl + #

Grid

Here you can set the grid type. To choose from:

- Object grid
- Grid range
- Beat grid
- Beat grid (relative)
- Frame grid

Detailed information on snap and grid settings can be found under "File -> Project Properties -> Snap and Grid Setup (view page 467)...".

Snap and Grid Setup

Detailed information on snap and grid settings can be found under "File -> Project Properties -> Snap and Grid Setup (view page 467)".

Show 2nd Grid Line

This command displays a second grid.

Exchange Grids

This command exchanges the upper and lower grids.

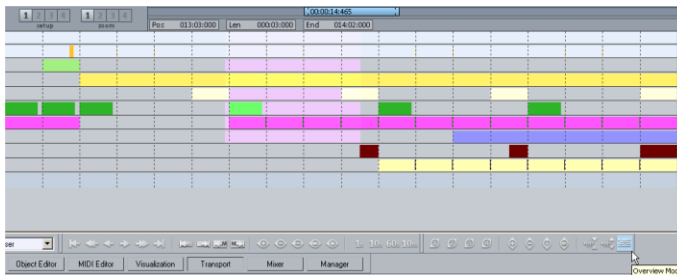
VIP Display Mode

- Definition...
- Mode 1
- Mode 2
- Toggle between mode

Detailed information on display options can be found via "File > Program Preferences > Definition > View Options (view page 491)".

Overview Mode

"Overview" mode displays all objects in the project clearly arranged below the arranger tracks. The mouse pointer becomes a magnifying glass in this area. The section selected in "Overview" mode specifies the size of the display and position section displayed in the arranger. You can position the section in the overview by clicking the desired location. On the other hand, the scroll bar in the arranger allows the selected section to be controlled in the overview.



To select the desired section, right click (or left click + "Shift") and drag out a lasso across the overview as required in the overview. If another position in the overview is required or a new lasso is drawn by left clicking, then the horizontal (i.e. temporal) position in your arrangement will change; the vertical range of the section (i.e. the display of the selected track) remains preserved.

In single-track virtual projects (VIP) and wave projects, the waveform will be shown in "Overview" mode.

Fix vertically

This menu item is hidden by default. To display it again, activate it in the menu settings via "File -> Program Preferences > Edit Keyboard Shortcuts and Menu (view page 488)".

If this function is inactive, i. e. the check to the left of the menu item is not set, then ranges in Samplitude's wave editor may be moved horizontally and vertically. In this mode, additional functions such as amplitude measurement for the De-clipper and the dynamics processors are available. Get level from area height: This function sets the DeClipping level

according to the height of the selected area in the wave project. This is useful when an overmodulated area has been previously selected exactly, and now has to be repaired.

If you have set "Vertical lock", the range display will include the complete wave editing window, like normal.

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

Horizontal

This menu contain horizontal functions for clip control, which are also partially represented in the bottom position bar as buttons.

Section to Beginning	
Section Left	
Half Section Left	Ctrl+Alt+Left
Half Section Right	Ctrl+Alt+Right
Section Right	
Section to End	
Section to Play Cursor/Last Stop Position	Ctrl+Alt+,
Section to Range Start	Ctrl+Shift+Page up
Section to Range End	Ctrl+Shift+Page down
Zoom In	Ctrl+Left, Up
Zoom Out	Down, Ctrl+Right
Show All	Ctrl+Alt+Up
Zoom to Range	Ctrl+Alt+Down
1 Pixel = 1 Sample	
Zoom Level 1s	
Zoom Level 10 s	
Zoom Level 60 s	
Zoom Level 10 min	
Definable Zoom Level 51	
Definable Zoom Level 52	
Definable Zoom Level 53	
Definable Zoom Level 54	

Shortcuts:

Half section left	Ctrl + Alt + Left arrow
Half section right	Ctrl + Alt + Right arrow
Cut to play cursor	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 play cursor in the current VIP's image section, then the command "Section to play cursor/Last stop position" moves the visible section to the current play cursor 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.

Section to Upper End	
Section Up	
Half Section Up	Shift+Up
Half Section Down	Shift+Down
Section Down	
Section to Lower End	
Zoom In Track	
Zoom Out Track	
Show All	
Range to Section	
1 Pixel = 1 Bit (-90dB)	
Zoom In Wave	Ctrl+Up
Zoom Out Wave	Ctrl+Down
Zoom default (0dB)	
Zoom In Volume Automation	
Zoom Out Volume Automation	
Zoom default Volume Automation	

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

Cascade

This function arranges all open windows one behind the other.

Tile

This function arranges all open windows one on top of the other.

Shortcut: Enter

Untile

This function arranges windows in the same way as they were prior to "tile vertically".

Shortcut: Shift + Enter

Window

Arrange Icons

Arranges the project icons along the lower edge of the monitor.

Close All Windows

This closes the currently opened project. A confirmation will ask if you want to save the project.

Shortcut: Ctrl + H

Wave Projects to Icons

Samplitude will display the wave projects as icons along the lower edge of the arranger window.

Hide All Wave Projects

Use this function to hide the wave project's icons and windows.

Half Height

This command displays only half of the Samplitude screen.

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.

1, 2, ...

Samplitude shows a list of the currently opened projects in the last section of the "Window" menu. A check shows which project is active. Select a project from this list to activate it in the project window.

Help Menu

Help Menu

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

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

Open Manual...

Use this command to open a PDF version of the Samplitude manual.

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 Wizard / Tip of the Day

This command opens Samplitude's start selection. This command may also be used to display the demo project, most recently opened projects, the "tip of the day", or the intro video.

The start selection may be used to add a new multi-track project, load an existing multi-track project, load a wave project, and access the recording options for a wave project.

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

Dongle Activation...

Detailed information about activation is provided in "Installation"

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