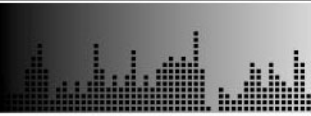
samplitude®

manual

the master of pro audio
recording - editing - mastering



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Support

If you experience any problems with your software, please get in touch with our support team. It is compulsory that you register Samplitude before using one of the offered support options. Help is also available at the official Samplitude website:

www.samplitude.com

This website will lead you to the Samplitude user service page containing, among other things, the following free offers:
- **FAQ**s (frequently asked questions) and general **tricks and tips**. In most cases you'll find the solution to your problem here. If not, use the support contact.
- **Support forum**: You are not alone. Perhaps other users had a similar problem and can help you solve yours. Our support staff is also a regular contributor.
- **Download section**: Updates, improvements and patches are likewise offered via download. Many problems you may experience are well-known to us, and can be solved by downloading the latest patch. Besides patches, there are also wizards for checking and optimizing your system.
- **Newsgroup**: here you can find a direct line to all other Samplitude users.

All Countries incl. USA:

You can also reach our support team either by email, telephone or fax.

Email: proservice@magix.net
Phone: +49 (0)351 4174616 (weekdays from 10 a.m. – 3 p.m. CET)
Fax: +49 (0)3514 796 2410

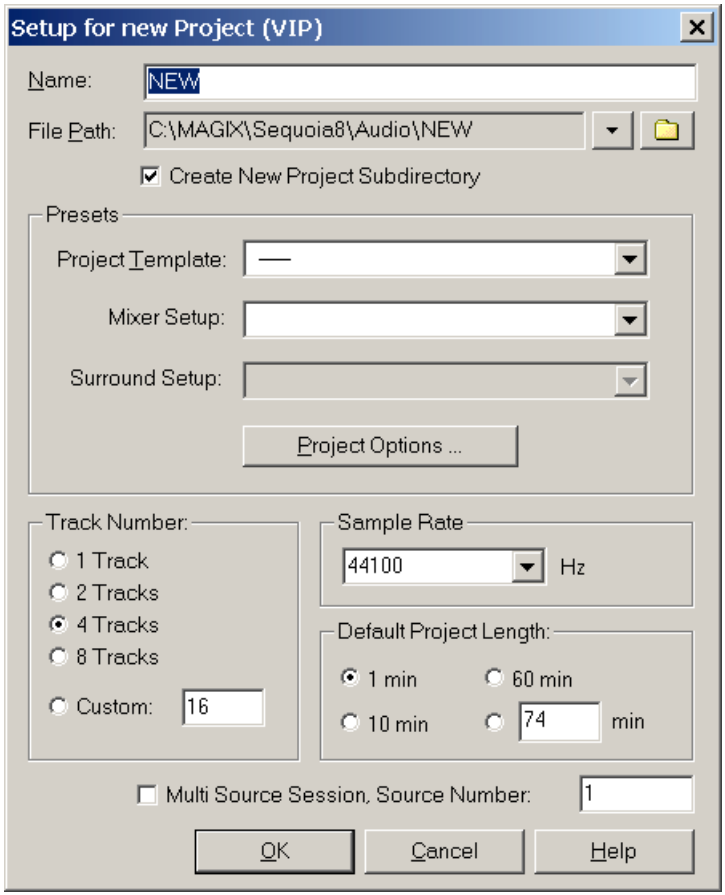
Please have the following information at hand:
- Program version
- Configuration details (operating system, processor, memory, hard drive, etc.)
- Sound card configuration (type, driver)
- Information regarding other software installed

For current price information regarding upgrades, etc. please contact your local dealer or distributor, or contact MAGIX AG directly on
Tel: +49 (0) 5741 345 525
Email: samplitude@magix.net"

“File“ Menu

New Virtual Project (VIP)

A new Virtual Project is created, which is immediately presented in a new window in the Samplitude workspace.



Project Template: You can choose between previously selected project templates in this list field. It includes all project settings such as track number, device assignment, etc. Use “Save Project as Template” to save project templates.

Mixer Setup: Lets you choose a predefined mixer setup. Mixer setups consist of number and type of the tracks or busses, and all input/output and effect routings. Read more in “Mixer Setup”. If you want to create a surround project you should set the format here. Read more in “Mixer Setup” and “Surround”.

Project Options: Opens the Project Options Dialog. All settings (e.g. Grid, BPM, CD arrange mode, autosave, etc.) made here apply to every newly created VIP. For this purpose they are saved in a special VIP (templates/template.VIP).

The "template.VIP" file can also be edited directly. You can then also make additional settings as default settings for your project:
– Record arming setting of the first track
– Some mixer setup settings
– Playback mode
– Track and master effects settings (e.g. compressor mode, effects folder)
– Grid offsets: Sets the Grid Start time in respect to the VIP start time.

Track Number: The track number of the Virtual Project is determined here. You can add tracks later using the "Track -> Add new tracks... " command.

Sample Rate: Defines the sample rate of the Virtual Project.

Note: Objects with nearly any sample rate, audio format or coding can be loaded irrespective of the VIP sample rate.

Name: The name of the new VIP should be included here.

Create New Project Subdirectory: A subdirectory bearing the name of the project is automatically created in the Virtual Projects folder (refer to Settings/ Paths). This is the easiest method of keeping all files which belong to a project consistent.

Default Project Length: The VIP is created with this length (1, 10, 60 min or a predefined value).

Shortcuts: *e*

Open Project → Virtual Project

A multi-track project in Samplitude. The dialog looks for files with the .VIP extension.
When loading a Virtual Project, all associated Wave Projects are opened, if they were not opened prior to loading the VIP. The windows of the individual Wave Projects remain minimized to prevent screen clutter. They are initially only visible as icons, and will be covered if the VIP is maximized.
When loading or recording .WAV files into the VIP window, each resulting Object has a time stamp, reflecting its original time position. Any VIP opens in a dedicated window in the Samplitude workspace.

Shortcuts: *o*

Open Project → MIDI (*.mid)

Allows you to import single or multiple MIDI files into a VIP.

Refer to the MIDI chapter for more details.

Shortcuts: *Shift + m*

Open Project → RAM Wave (RAP)

RAM Wave Project. RAM Wave Projects contain audio data in Samplitude's proprietary format. These files are loaded into the RAM of your computer, including their associated graphic files, marker position information, etc.

Shortcuts: *l*

Open Project → HD Wave (HDP)

HD Wave Project. HD Wave Projects contain audio data which is directly loaded from the hard disk, plus graphical information, marker position information, etc. These files are stored in WAV format.

Note: If a VIP is the active window, all loaded WAV Projects are immediately turned into objects in the VIP. Any selected Range's beginning determines the position and track where the Object is created.
An exception is, when the CD Arrange Mode from the Menu CD is set. The Objects are placed completely independent from any selected range. Instead, the Objects are placed in sequence (one after the other) with a pre-determined gap between them. The gap can be set with CD->Set Pause Time. The gap represents the space between individual tracks on a CD.

Shortcuts: *Ctrl + l*

Open Project → Object (*.Obj)

An Object which has been saved in this format contains playback instructions (link to a Wave Project, track, time position, parameters, etc.) for audio data. Objects are used in Virtual Projects.

Note: You can also use Windows Explorer to drag & drop saved Objects directly into VIP Projects.

Open Project → Edit List (*.EDL)

This function allows you to open an Edit Decision List in D-Vision Video Systems format. A Virtual Project that resembles the EDL is created. The list is a text file that contains information about the WAV files used, the Object borders, Volume, Panorama Automation, Mute, Lock and Fade information. This effectively makes Samplitude VIPs compatible with any application that supports this format, including EDL Convert Pro by CuiBono Soft. EDL conversion applications support the conversion to any other exchange- or project formats, including ProTools sessions, Sadie projects, AES31, OpenTL, OMF and many more.

Open Project → Table Of Contents (*.TOC)

Table of Content (TOC): Table of Content of a CD mastered in Samplitude. You can create .TOC files by using the command "Export TOC" from the "Make CD" dialog. When loading a TOC, the associated Wave Project, which contains

the raw audio data for the CD, is opened in a new VIP. In this VIP all CD track markers are set properly according to the CD.

Load audio file

Opens a file for loading audio files. The following formats are supported: Wave files (.WAV), MPEG files (.mp2, mp3, mpg), Apple files (.aif,.aiff), Sound Designer 2, Windows Media (.asf,.wma), Ogg Vorbis (.ogg), and Video for Windows soundtrack (.avi) and other video files (for MIDI files refer to Importing MIDI files).
You can open a AVI file to edit the audio track of the video directly (without importing the audio data from the video file). After finishing an edit, replace the audio in the Video with the "Export Video Sound" command from this menu.
The file formats .WAV, MPEG, OGG, AIFF and AVI are read directly by Samplitude; all other formats are imported when opened, i.e. saved on the hard drive as a new Wave Project.
You can also load several files simultaneously. To do this, expand your selection via Ctrl+Click or expand to a range of files via Shift+Click.
Note: Due to a peculiarity of Windows Explorer, the line in the file selection menu is laid out from back to front. Therefore, if you, for example, click on Track1, Track2 Track3 (using "Ctrl"), you will see "Track3", "Track2", "Track1" in the input field. (They are also loaded in this order.) So, if you want to load several individual titles, select them in the reverse order to which you would like them to be arranged in the project. However, if you select a range in Explorer (via Shift+Click), for example an entire folder, you should proceed as follows: First select the last title in the list. Then click on "Shift" and mark the first title of the list. If you now click on "Open", all tracks are in the VIP in the proper order.
If a VIP is opened, new Objects are created simultaneously in the VIP which refer to loaded audio files. If no area has been selected in the VIP, the files are positioned after the last Object.
More positioning options for several files should be adjusted in the "Options for loading audio files" menu.
Each audio file can be previewed before loading.

Note: The preview function uses the standard output device of the Windows Multimedia system (as do the system sounds). Many audio cards deactivate the normal Window MME Sound System when using the ASIO driver. This results in error messages in the preview function. A preview function when using the ASIO driver is first deactivated by default. However, you can activate it again ("Options for loading an audio file") if you select a different audio device for the multimedia functionality of your PC, for example, the onboard sound available on most modern PCs. Alternatively you may use the new File Browser function among the new Manager dialogues.

Key w

Load Audio CD Track(s)

See "CD Menu"

Save Project

The current 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.

Shortcuts: *s, Ctrl + s*

Save Project as

You can redefine the path and name of a new project with this function. RAM Projects and Virtual Projects are saved with the new names. (The source file remains as it was.) HD Wave Projects are renamed on the hard disk. Samplitude does not generate a copy of these, thereby saving space on your hard drive for Wave Projects.

Shortcuts: *Shift + s*

Save complete VIP in

This function in the "Project" menu allows you to save a VIP, along with all required Wave Projects (*.RAP. *.HDP), into a specified directory. This makes it easy to copy all files used in a VIP to, for instance, a backup disk.

Copy only samples used in VIP: Only the portions of Wave Projects that are actually *used* by Objects in the VIP are copied. This is the equivalent of using the "Remove unused samples" command (Tools menu) on the saved VIP after copying.
Use of this function can conserve (a lot of) storage space, but there's a catch. The downside is that after performing this function, Objects can only be made shorter, as all audio data outside the Object borders was removed. To circumvent this side effect, you can define a "security range" in samples. This number of samples will be left in the Wave Project - before and after the Object border – leaving a few samples in reserve, should you wish to change a fade in/out, for example. The default value is 22050 samples (= 500ms, at a 44.1 kHz sample rate).

Save Project as EDL

The active Virtual Project is saved as an Edit List in the EDL text format. The Edit List is a text file that contains information about the WAV files used, the Object borders, and volume and panorama automation information.
This effectively makes Samplitude VIPs compatible with any application that supports this format, including EDL Convert Pro by CuiBono Soft. EDL conversion applications support the conversion to any other exchange- or project formats, including ProTools sessions, Sadie projects, AES31, OpenTL, OMF and many more.

Save Project as Template

You can use this option to create project templates, which save all project related settings (grid type, track number etc.), without saving the Objects and HDP's used in the project. You can load templates when you create a new Multi-track project. This is a very convenient and useful feature which will speed up your work considerably and make it easier to maintain consistency across multiple projects. You should use this function and not derive new, empty VIPs from existing projects with the "Save as" command as different rules apply for the handling of projects paths for these two functions.

Burn Project Backup on CD

This option allows you to easily create data CD or DVD backups of whole projects. For this purpose, the external MAGIX CDR CD/DVD-burning application is launched. Please refer to the mxcdr online help for more information.

Save Object

This function allows you to save a single Object. This is especially useful where you wish to transfer material between VIP Projects, or would like to preserve certain parts of a VIP (by saving them to single Objects). Another use is the creation of a proprietary sound effects library.

Note: You can use Windows Explorer to drag & drop multiple Object files into a VIP for easy access.

The default file extension for Object files is **.obj*.

Note: It is important to ensure that only one Object is selected for saving. If more than one Object is selected, the first selected Object in the lowest numbered track (i.e. track 1) will be saved.

Save Format
Sessions in Samplitude are a quick way to restore the screen layout to the way the windows and toolbars were situated when the session was saved. Let's assume that you have a VIP open, the Mixer window, and several HDP windows. When you save a session and reopen it, Samplitude will reposition all open windows at the exact place where they were positioned at the time the session was saved.
In addition, if you save a session with the name "startup.sam" and place it into the folder with the Samplitude program file (default installation would be c:\MAGIX\Samplitude ...), Samplitude will load that session the next time you start the program.

Rename Project

This function allows you to rename a project file, rather than save it to a different file. For RAM Wave Projects, only the internal names are changed (without

being saved). In the case of HD Wave Projects, all corresponding files are immediately renamed. RAM Wave Projects need to be saved after renaming the project.

Delete Wave Project(s)

Wave Projects are deleted from the hard disk.
Use this command with caution, as all corresponding files are lost!!!
There is no undo for this function, and no recycle bin.

*Note: If you want to delete a Wave Project from a file manager such as Windows Explorer, you should also delete the related Samplitude files (i.e. *.hdp, *.ho, etc.)*

Delete Virtual Project

This command deletes a VIP, and all Wave Projects used in it.

File selection: You can select the project file (.VIP) to delete.

Project list: You can choose from a list of the eight most recent projects.

File list: The file list shows the Wave Projects used in the selected .VIP. You can specify - on an audio file by audio file basis – whether or not you wish to keep the file (if used in other projects), or to delete it.

Convert Audio

This submenu contains various options for converting mono/stereo Wave Project formats.
These conversions can be performed in Wave Editing mode, or by selecting a VIP Object prior to performing the command. In both cases, the original file is not changed – new files are created, and named, automatically by Samplitude.
Performing the command from within the VIP results in changes to the Object which reflect the conversion. In other words, the Object will now refer to the converted file(s).
Performing the command via Wave Editing (Destructive Editing) does not alter the existing VIP – if the resulting file/s are needed in the VIP, this must be done manually by opening the object editor (position/fades window) and exchanging the underlying Wave Project.

Convert Audio → Import Audio

This provides the possibility to import all supported file formats explicitly as Wave Projects. This is only necessary for file formats that cannot be opened directly by Samplitude (Windows Media Files and certain MPEG files).
However, you can increase the performance of your computer if you import compressed formats such as mp3 or Ogg instead of loading them in order to avoid real-time decoding.
Unlike "Load Audio File" the audio file is converted to Wave when importing and copied to the hard drive. You can also use the command if you have good reasons to work in a copy of the file within the project.

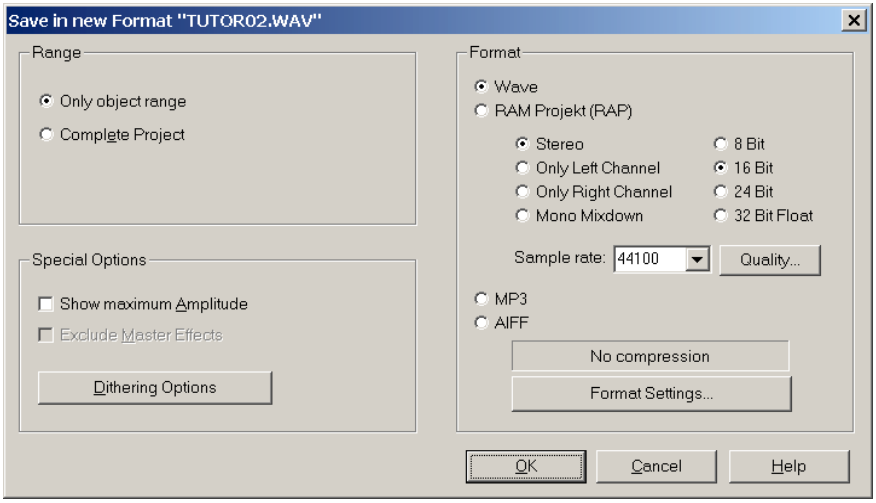
Key: *Ctrl + I*

Convert Audio → Save In Format

This function only operates when working directly on a Wave Project, and allows you to convert projects between the various Samplitude Wave Project formats. Another option is the conversion to/from stereo/mono Wave Projects and changing sample rate and bit resolution. Conversions into MP3 or AIFF format is also possible.

Range: Save the selected range or the whole file

Special Options: Show maximum amplitude: Enables display of the maximum level after saving in dB for correct adjustment of the master level.



Exclude Master Effect: The master effects are not included during save. Master effects are available for Wave Editing only in virtual Wave Editing mode (refer to the "Wave Editing" chapter) when you open the mixer window for a Wave Project.

Dithering Options: When reducing bit resolution from 24bit or Float to 16bit, dithering takes place. You can choose the algorithm used in the according dialog (see Options ->Program Preferences->Dithering Options).

Format: This section allows you to choose the new format specification. Select the desired format, and press OK to convert the audio file. You will be prompted for a new file name. A new file will be created, leaving the original audio file intact.

QUALITY: Choose the quality of the used resampling algorithm (See Options ->Program Preferences->Resampling Quality Options) when the sample rate is changed.

Note: It is not possible to use the same name and overwrite the existing file, as the file is actually open. (Samplitude will not overwrite an open file in this situation). You will then be asked to specify a name and location for the new file.

Convert Audio → Stereo Wave → 2 Mono

A stereo Wave Project is copied and converted into a discrete left and right mono Wave Project (i.e. the same sample is placed in each mono project).

Convert Audio → 2 Mono → Stereo/LR Wave

Two mono projects are linked to one L&R Wave or stereo project. This is a convenient way of editing joined mono samples with the same operations.
After sending the command, a dialog opens which displays all loaded mono files. "Load file" allows you to load other files. Choose the right and left files from opened Wave Projects using the ˆ key. The <-> keys let you change the channels. After clicking "Link Files", name the new stereo file which will then be created in the project directory.
Take note that mono files can only be joined with mono Wave Projects and that only projects with the same bit resolution and sample rate can be joined. The length of the two projects are then fit to each other; meaning, the longer mono file determines the total length.

Convert Audio → LR Wave → 2 Mono

On occasion, it may be necessary to split a stereo project or two joined mono projects (a so-called L&R Wave Project) into two independent mono projects. This menu option terminates the static connection between the projects. If you would like to re-join the projects, simply select 'Convert Audio > 2 Mono > Stereo' from the same menu (see below).

Convert Audio → LR Wave → 1 Mono

A current L&R Wave Project is converted into mono. If it was previously a stereo Project, both channels are mixed. The existing samples are initially added with 100% of their image intact and are then divided by two. This prevents clipping, and is equivalent to reducing the level by 6 dB.

Convert Audio → 1 Mono → LR Wave

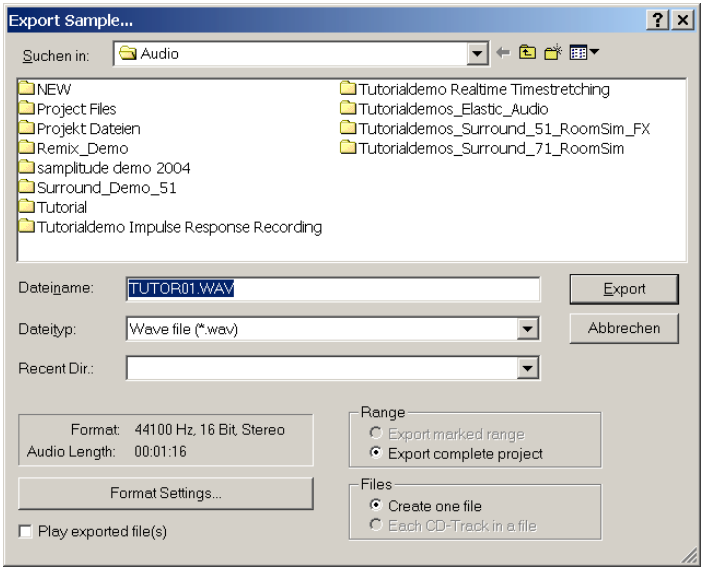
The original mono Wave Project is duplicated, and converted into a single L&R Wave Project, featuring the same sample in both channels.

Export Audio

The command "Export Audio" will let you export Wave Project files or VIP files into various audio formats.

A Note about exporting Wave Projects: Wave Projects can be directly loaded, or imported, as .WAV files by other audio applications. The Export Sample command is only necessary if you wish to copy the audio file or convert it into different formats. Keep in mind that capacity on your hard disk is required, and that the copy process takes additional time.

You can also export Virtual Projects via this menu command. The dedicated track bouncing command from the Tools menu offers much more control about the details of the bouncing process including multi track bouncing, exclusion of effects, bouncing a single track or Object of the track, for instance. The following export formats are possible:
Wave, MP3, MP3 with external encoder, MPEG, Windows Media, Real Audio, Ogg Vorbis, AIFF or AIFF with QuickTime, 32 Bit Float as 16/20/24 Bit and dump files.
You can also export MIDI Objects as Standard MIDI Files.



Options

The two radio buttons in this section allow you to export the entire file, or a portion of the file, selected as a Range.
You can furthermore decide, if you want to save each CD track of a project as a separate file or if you want to save the whole project as a single audio file. If the tracks are saved separately, an additional list file (. m3u) will be created besides the audio files. This file contains the names of the single audio files in the correct order. The option is only available when any CD tracks are set in the project.
To automatically play the exported file on completion of the process, check (tick) "Play exported file". The playback will be not be done by Samplitude but by the default playback application for this file type.

Format Settings button
Depending on the exported file format the button launches various format settings dialogs.

Export Audio → Wave

The main reason for exporting Wave Projects is to compress the file into a particular .WAV format. Samplitude can use various Windows .WAV compression codecs that are available in the system. Select your desired setting, and click OK.

Export Audio → MP3

This dialog enables you to export files in the MP3 format, with various format settings.

Export Audio → MP3 with external Encoder

You can invoke an external command line encoder, and can set several parameters for the encoder in the dialog.

Export Audio → MPEG

This dialog allows you to export files in the MPEG2 format (*.mpg). Clicking on the *Format Settings* button launches the *Choose Bitrate* dialog, shown below. Samplitude offers MPEG2 Bit rates ranging from 64 kbit/sec to 384 kbit/sec.

Export Audio → Windows Media (*.wma)

This dialog allows you to export files in the Windows Media format (*.wma)
Clicking on the *Format Settings* button launches the *Windows Media Export Settings* dialog, shown above.
Various MS Audio Bit rate settings are available, ranging from 6kbps for Voice Audio to 128kBps for near CD Quality Audio. The dialog also allows you to enter Title, Author, Description and Copyright details.
Redirector files (*.wvx) can be generated by following the directions in the *Redirector Files* section.

Export Audio → OGG Vorbis

Exports the project in OGG Vorbis format. Simply enter the filename in the dialog. This format is an excellent alternative to the well-known MP3 format.
The Format Settings button opens a compression settings dialog. The compression scale ranges from 46 kBit/s to 500 kBit/s (32 kBit/s to 320 kBit/s for MP3). No only can compression be adjusted in relatively large steps (e.g. 64 kBit/s, 96 kbit/s, 128 kBit/s, 160 kBit/s) but also in steps of 2 (e.g. 148 kBit/s, 150 kBit/s, 152 kBit/s).

Export Audio → Real Audio (*.ra)

This dialog allows you to export files in the Real Audio format (*.ra)

The initial dialog is identical to the *Export Audio > Wave* dialog.
Clicking on the *Format Settings* button launches the *Real Media Export Settings* dialog, shown above.
Various Real Audio encoding algorithms are available via use of the appropriate settings in the *content settings* and *target* menus.
Author, Title and Copyright information can also be entered.

Export Audio → AIFF File with QuickTime

This dialog allows you to export audio as Macintosh AIFF format (*.aif), through use of QuickTime's conversion algorithms.
The browser window, that opens after selecting the function from the menu, allows you to name the file and select a target directory for it. QuickTime will initially create temporary files in this directory as part of the conversion process.
The dropdown menu allows you to select various formats for the conversion process.
Further variables are available by clicking on the Options button.
The *Compression* dropdown menu allows you to determine the bit reduction / compression algorithm to use – this is, to some extent, dependent on the input file format.
The *Rate* dropdown tab allows you to select the desired sample rate.

Export Audio → Aiff

This menu allows you to export audio material as AIFF files, at the selected bit depth.
16 bit material can only be exported to 16 bit AIFF. To export 16 bit audio to higher resolution formats, you must first convert the file to 32 bit Float format with the *File > Convert Audio > Save in Format* function.

Export Audio → 32 Bit Float as 16 / 20 / 24 Bit

This menu item allows you to export 32 bit Float audio files in reduced bit depth .WAV formats.
Dithering is employed, providing the best possible quality at the lower bit resolution.

Note: If Samplitude's dithering functionality is deactivated, the sample will be rounded mathematically during bit reduction. This is a better solution than pure truncation.

Export Audio → Export MIDI File

This menu item allows you to export all MIDI data from the project as standard MIDI file (*.mid).

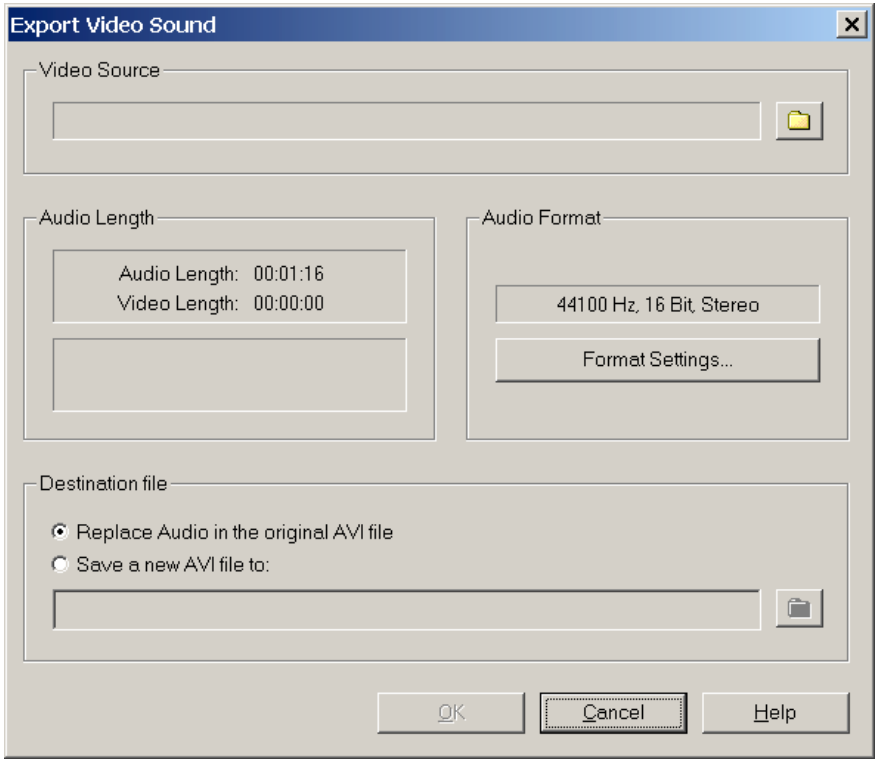
Export Audio → Export as Dump

This menu item allows you to export audio material as a dump file – only the raw sample data is stored in the file. (i.e. no file headers, etc.)
You can choose between "Intel" and "Motorola" byte order. (So called Little- or Big Endian.)

Note: You must convert 32 bit Float files to 16 or 8 bit before using this command. Only 16 bit audio files can be exported as Dumps.

Export video sound

You can load video files like audio files to work on the video soundtrack. This is used, when you don't want to edit the audio, and just want to process the whole audio track with effects, like normalizing or noise reduction. If you prefer a visual feedback, use the function Menu Options > Media Link instead. After editing the video sound track, you can re-write sound track back into the video file. You also have the possibility of replacing either the original video sound track or of producing a new video file. a new time-consuming compression of the video is not necessary. No matter what, you can change the audio format, for example, into compressed formats such as ADPCM. However, the appropriate compression codecs must be installed.



Create AVI / Audio Copy: This option performs a track bouncing of the current VIP into an audio file. The AVI file is copied to a new file, which contains the new audio track. This option keeps the original AVI.

Replace Audio in Video: This option performs a track bouncing of the current VIP into an audio file. This file is merged into the selected AVI file, so the AVI gets a new audio track.

Note: If the length of the video file and the audio are different, a warning is given. After exporting, the longer one is truncated, no automatic synchronization is done. If you obtain such a warning, try to resample your audio soundtrack to the appropriate length.

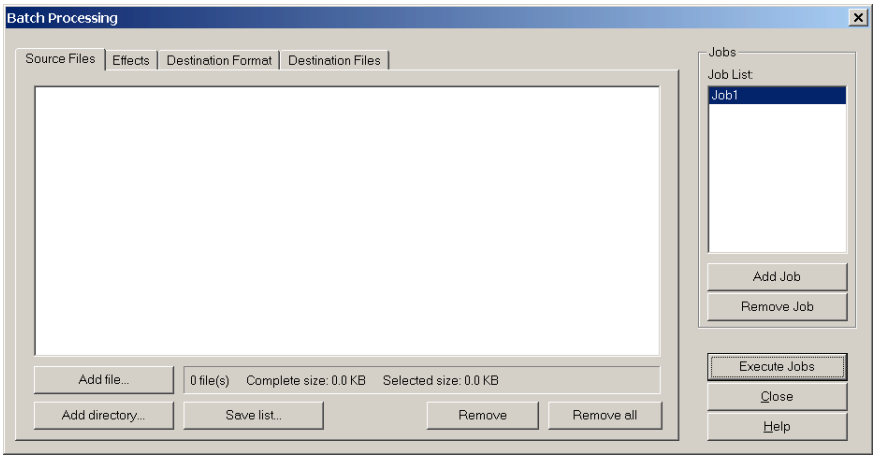
Export to Video deLuxe pro

The Menu command calls the track bouncing dialog and opens the MAGIX video deLuxe video editing program with the result.

Batch Processing

Batch processing automatically applies a single Wave Project task to any number of other files (the "batch") in exactly the same way – for large volumes, usually overnight.
Each task ("job") is entered in a job list that defines several batch jobs They can define several batch jobs which are then implemented one after the other.
Possible jobs may include:
- normalization
- linear fading (in and out)
- real-time effects in the mixer master
- Format conversion: Bit resolution(16/24/32), Sample rate, stereo/mono/left/right
- Storage in all available export formats

For example, you can normalize a whole list full of 24bit .WAV 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 16bit mono, resample them on 22kHz and then saved them all as mp3s.



Jobs

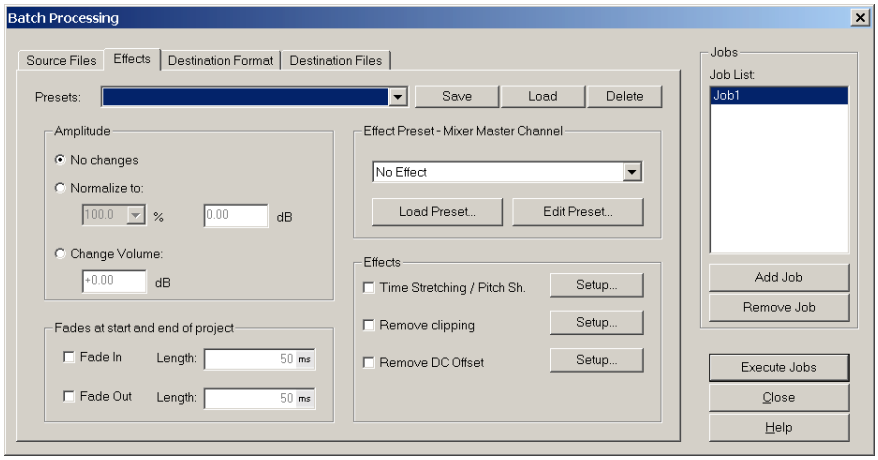
To the right edge of the batch processing window is the job list. New jobs are created by activating the "Add job" button. When you select a job (with a mouse click), the 4 settings tabs are displayed(source files, effects, target format, target

files). "Remove job" removes the selected job. "Perform jobs" starts the batch processing.
As long you don't close down Samplitude, all the jobs remain saved until execution, even if you close the batch processing window in the meantime.

Source files

Here you can create a list of the files to be edited by selecting "Add file". Multiple selection is possible. Select "Load directory" to add all of the audio files within a directory (including all sub-lists). All importable audio formats are loadable in Samplitude. "Save list" creates a Playlist (in the common *.m3u format) for using your selection of files later. "Remove" deletes all selected list entries. " Remove all" deletes the complete list.

Effects



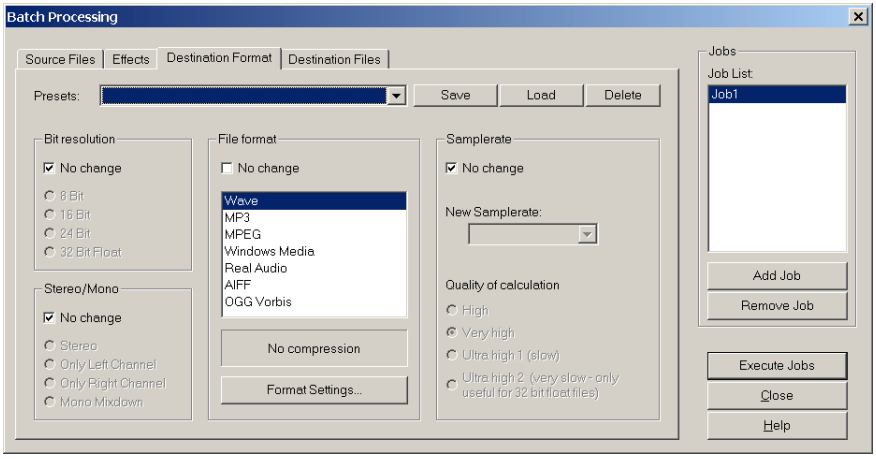
1. Normalization
You can normalize using percentages (%), decibels (dB) or manually by using the levels. For more details, please read the Effects Menu chapter -> Amplitude/ Normalize

2. Master Effects
To avoid redundance in the batch processing dialog, effects settings have been integrated with the mixer preset files.
{ bml buttons_mx_save_effect.bmp} In the Samplitude mixer, you can export all mixer settings to the mixer preset (*.mix). Simply press the "Save mixer preset" button in the mixer (see illustration).
By loading these *.mix files into the batch processor ("Load preset"), all master channel effects settings are included (all other mixer preset data e.g. channel properties, groups, etc. are ignored). "Edit preset" opens a special FX routing dialog from which you have access to all effects parameters.

For more details about the FX routing dialog, please read the "Effects" chapter. Mixer presets are stored in the /fxpresets/batch directory and can be selected directly from the listbox.
Besides mixer preset master effects, you can use the non-real-time effects Resampling/Timestretching, DC removal and De-Clipping. Settings opens the applicable effects dialog.

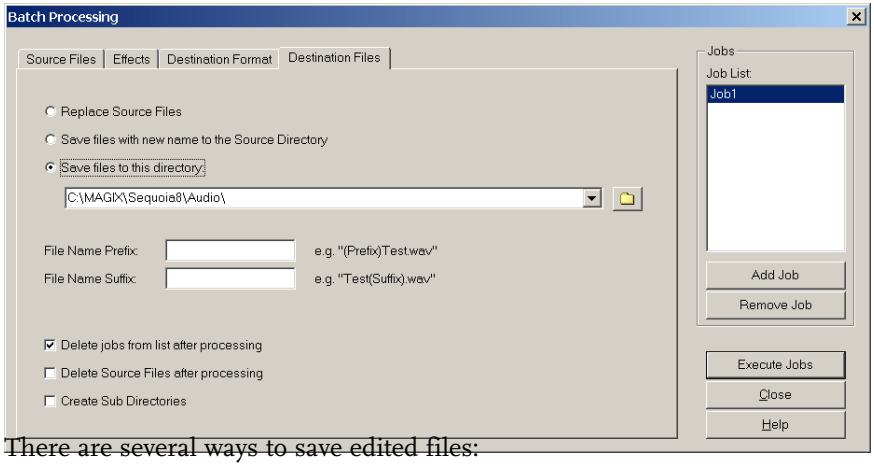
3. Fades at the start and end of projects
Linear fades of any given length can be added to your files.

Destination format



All settings can be saved as a preset. For more details, please read Samplitude Effects -> Saving Effects (Preset mechanism).
You may select the bit resolution(16/24/32 bit), the sample rate, stereo/mono/left/right and format with the applicable format option.
For more about the sample rate: Effects menu -> modify sample rate .
For more about export formats: Effects menu -> export audio.

Destination files



There are several ways to save edited files:
"Replace file": Replaces the original file with the edited file. Should the file be used in the VIP, the VIP first must be closed.
"Save and rename file in the source directory".
"Save file in the following directory".
"Delete source file after editing".
"Save with source directory path". The file is saved within the path to its source.

Connect to the Internet

Opens your default dialup connection to the Internet. All you need is a normal internet-connection for your computer – using a Modem, ISDN or ADSL.

FTP Download

This option allows you to directly connect to any FTP-server on the internet, in order to download audio material into Samplitude. The MAGIX server is the "default", which is not a bad thing, as you'll always find new sound-material there. Once again, the only requirement is a working internet connection.

Close Project

This menu command closes the active project window. If the project is changed, but not saved, a warning message will pop up.

Exit

Closes Samplitude.

"Edit" Menu

This menu contains all functions that are analogous to cutting/splicing etc. tasks that you would perform on a tape machine.
Virtual Projects are "track-sensitive" – i.e. only the selected tracks are manipulated. When working in a Virtual Project (VIP), you must first select a Range before making use of the cut functions. The start and end points of the selected Range determine the start and end points of a cut. The vertical position indicates the tracks that are affected by the edit.
We recommend the use of the Auto-crossfade function for almost all situations, in order to achieve smooth transitions between the edit points. This will automatically create fades at the Object edges.
Please note that these functions affect both channels, when working with stereo and L&R Wave projects. (If only one channel is to be edited, you will need to convert the Wave Project into two mono Wave Projects. The two mono Wave Projects can be merged, following individual processing).
Attention: To avoid inconsistencies between the Object modes (e.g. lock all audio in time, link Objects until silence) and the common edit commands, some VIP edit options work differently to those of former Samplitude versions.
General Rule: In VIPs, delete, cut and paste commands are dependent on the current Object mode.
Explanation: There are several Object "modes" (link Objects one/all tracks, link until silence), which "ripple". To explain, this means that when you move one Object in time, the following Objects are also moved, thus preserving crossfades and relative distances between the Objects. In these "rippling" Object modes, the edit command functionality is also affected. To give you an example, when you cut one Object, the following Objects will be moved backwards in time, to fill the gap. If you perform an insert, the following Objects are moved forward, to avoid overlaps. This is the "standard" behavior of the delete, cut and paste functions, as in former versions of Samplitude.
In other Object modes (normal, lock all audio), these commands will not perform a "ripple". In other words, temporal (time) movements of other areas not directly involved in the cut, remain unaffected. This facility is especially important when cutting time-synced audio, e.g. projects with .avi links. As an example, in the "normal" Object mode, the delete command works like the clear command, leaving a gap where the audio material was deleted.
The usual commands, which work independently from the selected Object mode are accessible in the "more...." submenu. If you find this behavior uncomfortable, remap your keyboard shortcuts to the old style.

Undo

Undo last command(s) (VIP). Samplitude offers a secure way of tracking your changes in Virtual and Wave Projects. Up to 100 changes can be kept in the memory, allowing you to retrace (and undo) all of these steps, if necessary.
Virtual processes and physical processes (such as Normalize, Reverb, Filter with activated Create Copy option etc.) can be reversed (undone).

Range and Marker manipulations can also be undone using this command. When adding effects to audio material (Effects Menu), the Undo option only works on effects that are applied directly to the VIP Objects – i.e. destructively "printed" to the Object, as opposed to realtime effects. The 'Create Copy' option must be enabled in the effect dialog, in order to "undo" destructive effects processes.

Notes: The number of Undo steps (i.e. Undo depth) is defined in the Options > Program Preferences > Undo Definitions Menu item.
Some commands, such as Tools > Remove Unused Samples or CD > Make CD, do not allow Undo commands.

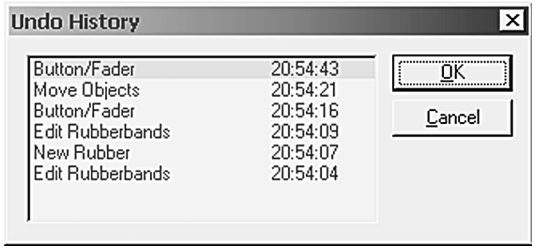
Shortcut: *Alt + Back, Ctrl + z*

Redo

Reverses (undoes) the most recent Undo command.

Shortcut: *Ctrl + a*

Undo History



Displays a list of recent editing steps. To return the project to the desired state, simply select the relevant Undo step. Note that it is easy to toggle between various steps, to ensure that you select the correct state. Once you're sure of your selection, click OK to complete the Undo.

Cut

Wave Projects:
The audio data (in the selected Range) are copied from the current Wave Project into the Clip. The material which follows the removed Range is merged with the material which preceded the removed Range, to close the gap. The complete Wave Project becomes shorter.
Please keep in mind that the Clip contains the same attributes as the Wave Project that the material was copied, or removed, from.
If you cut material from a mono Wave Project, the Clip becomes a mono Wave Project. If material from a stereo Wave Project is cut or copied to the Clip, the Clip becomes a stereo Wave Project. Another attribute adopted by the Clip is the bit resolution and sample rate of the source Wave Project. The previous contents of the Clip are deleted.

Virtual Projects (VIP):
Object Mode: Normal, Lock all audio
Cut works without ripple, the range is filled with silence, not moving following audio material backward.(=Command "Copy & Clear ")
Object Mode: Link Objects one/all tracks
The material behind the removed range is appended to the material in front of the removed range.
The VIP becomes shorter if the selected range covers all tracks. This way you can shorten VIPs which contain empty space behind the last Object. (= Command "Paste with Ripple")

Use the function Edit->Paste/Insert to insert the Clip contents in Wave Projects or VIPs.
Shortcuts: *Shift + Delete, x, Ctrl + x*

Delete

The command deletes all selected Objects, curve points or the selected Range in the project. Curve points have priority over ranges, the Range has priority over a selected Object.
When both an Object and a Range is selected, the Range will be deleted. That's why if you want to remove an Object from the project with this command (by using the Del key) a Range must not be selected.
A dedicated "Remove Object" command which works independently from a Range selection can be found in the Object->Cut Objects menu)
In accordance with the selected Object mode in VIP's, deleting a Range works as follows:

Object Mode: Normal, Lock all audio
Delete works without ripple, and does not move the ensuing audio material backwards (=Command "Clear").
Object Mode: Link Objects one/all tracks and Wave Projects
The sample data that follows the deleted Range is added at the position the deleted Range started (moved backward). The Project length becomes shorter (=Command "Delete with ripple"). Using this 'Delete' command will only delete the data from the current sample, leaving the VirtClip and its contents as they were. If you want to preserve the deleted material, use the 'Cut' command.

Shortcuts: *Delete*

Copy

Copies the selected Range into the Clip. (VIP, Wave Projects)
The current Range is copied into the Clip, but is not deleted from the Project. The sample length is unchanged. Please note that any previous Clip contents are deleted.

Shortcuts: *C, Ctrl + C, Ctrl + Insert*

Paste/Insert Clip

The contents of the Clip are inserted into the current Project, at the current play cursor position, or at the start point of the currently selected Range.

Object Mode: Normal + Lock all audio
Paste works without ripple, and does not move the subsequent audio material forward. This avoids overwriting/overlapping the material in the selected Range (= Command "Overwrite with Clip").

Object Mode: Link Objects one/all tracks and Wave projects
The data or Objects positioned after the insert point are removed to make room for the Clip contents. The samples or audio tracks become longer. The Clip remains unchanged during the procedure. (=Command "Paste with Ripple")
In Virtual Projects with selected Ranges, the Range start position, and relative track position, are used as the insert point for the Clip contents. Right after the insertion of the Clip, the program selects a Range over the inserted area. If you next select Edit > Delete , the inserted contents are removed, and the project is returned to its pre-paste state.

The following table indicates Samplitude's response in cases where the Clip and the Project have different channel numbers:

Clip >	Project	Clip Channel --->	Project Channel
Mono	Mono	Channel 1	Channel 1
Stereo	Stereo	Channel 1	Channel 1
		Channel 2	Channel 2
Mono	Stereo	Channel 1	Channel 1
		Channel 1	Channel 2
Stereo	Mono	Channel 1	Channel 1

Shortcuts: *V, Ctrl + V, Shift + Insert*

Extract

Deletes sample data on both outsides of the selected Range.
This function is designed to be the counterpart of the Cut function.

HD Wave Projects:
The current Range remains unchanged – the sections on either side of the selected Range are permanently deleted. The audio file becomes shorter. The contents of the Clip are unchanged.

Virtual Projects (VIP):
The selected Range determines the material (from all VIP tracks) that remains in the Project, after using the function. This is regardless of the selected Range

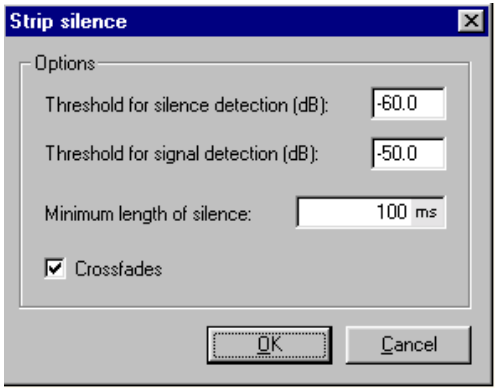
spans all tracks. In other words, the function is not reliant on specific track selection.
The Objects on either side of the selected Range are deleted from the Project. The contents of the VirtClip remain unchanged.

Insert Silence

The 'Insert Silence' option will insert 'blank' data at the current play cursor position, or at the start position of the currently selected Range.
You can enter the length of inserted silence in the displayed dialog, if a Range is selected, it's length is automatically used in the dialog as default value.
The "blank" space will actually contain data with a value of zero and will be inserted before the selected Range. Any data which follows will be appended to the end of the blank space. The defined Range is maintained, and the length of the inserted space extends the overall sample length.

Strip Silence

This command removes the silent parts from all selected Objects in a VIP. After application of the command, the Objects were split in a way, that the silent parts of the Objects are separated from them. The new silent Objects are all selected, so you can delete them all at once by pressing the DEL key. Or you keep them and examine them first.



Append Projects

This function allows a project to be appended to another project, i.e. the material of one project is copied, and placed directly behind the material of the first. You first need to select the Object you want to append, select the menu option, and click on the project you want to add to the first.
This option is commonly used for 'cleaning up' VIP's that contain numerous Wave Projects. You can:

- append a VIP with a VIP. This is good for bigger projects, you can edit parts of it in different files and finally put them together
- append a Wave Project with a Wave Project. A particular use of this option is the 'cleaning up' of VIP's that contain numerous physical samples.

– append a Wave Project with a VIP. This is a smart method to transfer markers from a Wave Project to the VIP. Normally, when a Wave Project is loaded into a VIP, the markers stay inside the audio file as Audio Markers. Often it is necessary to have these markers in the VIP. When you append a Wave Project with the VIP, the Wave data will be inserted as an Object at the end of the VIP at the active track and the markers from the Wave Project will be inserted in the VIP.

More → Delete with Time/Ripple

This command deletes the marked Range. Material later than the end of the Range will be preponed by the length of the Range.

Shortcuts: *Ctrl + Delete*

More → Clear

Replaces the selected Range with silence (VIP). This command is useful for removing material without changing the length of the VIP track(s).

Note: This command will not save the deleted data to the Clip. If you want to preserve the deleted data in the clipboard, use the Edit > Copy+Clear command.

Shortcut: *Alt + Delete*

More → Copy + Clear

Copies the selected VIP Range into the Clip, and clears it from the VIP Project. The Project length is not changed. Please note that the previous Clip contents are deleted.
This is a good way to remove a section of a VIP for use elsewhere, without affecting the length of the track(s) that the material was removed from.

Shortcuts: *Alt + C*

More → Cut with Ripple

The material behind the removed Range is appended to the material in front of the removed Range.
In VIPs: The VIP becomes shorter if the selected Range did cover all tracks. This way you can shorten VIPs which contain empty space behind the last Object.
Shortcut: *Ctrl + X*

More → Copy as...

Copies the selected Range into a new file. *(HD Wave Projects only)*
An Explorer-type file browser is launched, allowing you to name the new file, and select a target location for it.

Shortcut: *Shift + C*

More → Paste with Ripple

The samples or Objects that are located behind the insert position are postponed to make room for the Clip contents. The samples or audio tracks become longer. The Clip remains unchanged during the procedure.
Shortcut: *Ctrl + V*

More → Overwrite with Clip

Overwrites sample data with the contents of the Clip.
The current Range is replaced with the Clip contents. The overall project length remains unchanged.

Note: In VIP's, this command does not overwrite the actual audio data i.e. the HD Wave Project does not change.

The Clip contents are not changed.
In Virtual Projects, the selected Range determines the target position and track for the Clip contents.

Shortcut: *Alt + V, Insert*

More → Crossfade with Clip

The content of the Range is crossfaded with the content of the Clip. In this case the position of the play cursor determines the end of the clip which is to be used to crossfade the Wave Project. The length of the clip determines the length of the crossfade.
Also here, the rules of channel ordering are observed between Project and Clip in the same way as for the Paste function. The Clip remains unchanged.

More → Mix with Clip

This command allows you to mix sample data with the Clip. (HD Wave Projects only)
The contents of the Range and the Clip are evenly mixed. Channel assignment(s) between project and Clip follows the given pattern. The contents of the Clip are not altered. As both components are combined with their full sample values, make sure that no clipping occurs. The Mix with Clip function is performed by addition, ensuring that the Project sample remains free of sudden volume decreases.
Despite this, however, you may still need to modify the amplitude, before performing the mix, in order to keep the resulting sample from clipping / distorting. This can be done through use of the *Effects > Amplitude/Normalize* menu item.

Append Projects

This function appends one project to another - i.e. the material of one project is copied directly to the end of the first.

You first need to select the project you want to append another to, select the menu item, and click on the project you want to add to the first.
You can:
– append a VIP to a VIP. This is good for bigger projects, allowing you to edit portions of it in different files, and join them together.
– append a Wave Project to a Wave Project. A common use of this option is for "cleaning up" VIPs that contain numerous physical samples.
– append a Wave Project with a VIP. This is a "smart" method, used for the transferal of markers from a Wave Project to a VIP. Normally, when an Wave Project is loaded into a VIP, the markers stay inside the audio file as Audio Markers. Often, it may be necessary to have these markers in the VIP. When you append a Wave Project to a VIP, the wave data will be inserted (as an Object) at the end of the active track in the VIP, and the Audio Markers will be inserted into the VIP.

Auto Crossfade Active

If this option is active, 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/out applied to it. If you perform a "Split Object" command, the resulting Objects are crossfaded, avoiding clicks at the beginning and end of the Objects.
Global settings for Fade In and Fade Out parameters are assigned to the Object. These settings can be changed in the Edit > Crossfade Editor, which may prove useful where the audio material is cleanly cut, and the fade impairs the attack phase of the sample.
If two overlapping Objects have an automatic crossfade applied to them, a real-time crossfade results. If an Object with an existing fade in is moved over an Object that precedes it, a crossfade will be applied across the two Objects (when auto crossfade is active).
The Auto Crossfade mode is an excellent aid when making linear cuts to a spoken voice track, jingle, etc., which requires smooth segues - without those unwanted pops or clicks. If needed, each crossfade can be edited in the editor, or by manipulating the fade handles directly in the Objects.

See the "Crossfade Editor" chapter for more details.

Delete Curve Handles

The options offered by this command allow you to delete automation handles from the active VIP, as follows:

Delete Curve Handles → Delete Volume Handle

This function deletes any selected Volume curve handles. Individual Volume events can be deleted by selecting the Object in Curve Mouse Mode and then double-clicking on a Volume curve event. When deleting several curve events, the 'Delete Volume Handle' function is the best tool for the job. Use the Curve Mucse Mode, and grab tool, to select multiple curve events.

Delete Curve Handles → Delete Panorama Handle

This function deletes any selected panorama curve events. Individual events can be deleted by selecting the Object in Curve mouse mode and then double-clicking a Pan curve event. When deleting several curve events, the 'Delete Panorama Handle' function is the best tool for the task. Use the Curve Mode and Grab Tool, to select multiple curve events.

Delete Curve Handles → Delete Surround Handle

This function deletes any selected Surround curve events. Individual Surround events can be deleted by selecting the Object in Curve Mouse Mode and then double-clicking on a Surround curve event. When deleting several curve events, the 'Delete Surround Handle' function is the best tool to use. Use the Curve Mode and grab tool, to select multiple curve events.

Delete Undo Levels

Deletes all Undo steps (VIP). This feature will delete the Undo steps of a Virtual Project. This is sometimes useful if Wave Projects are left in the Undo chain, but no longer exist in the VIP. You won't be able to process or delete the Wave Project if the undo chain still contains a reference, so you might as well get rid of it. After deleting these "orphaned" Undo levels, these Wave Projects will, once again, be available for use.

"View" Menu

The View Menu allows you to adjust various parameters that determine how Samplitude displays project data. A wide variety of options are available, ensuring that Samplitude's display suits your preferred working methods, by selection of view options appropriate to specific workflows.

Rebuild Graphic Data

Occasionally, you may encounter inaccuracies in Waveform displays (or other project bitmaps), following extensive sample processing tasks. Such artifacts can be caused by a number of factors, including inefficient display drivers. Should this (rare) problem occur, simply use this Menu item to restore the graphic display. The screen (window) is cleared, and completely redrawn by this function.

Note: Corrupt graphic files can also result from abnormal program termination – i.e. a crash / hang. Corrupt graphics files are usually not a problem for Samplitude, as the graphics are not part of the audio data. Any corrupted graphics files will automatically be rebuilt when Samplitude is restarted. The project is then reopened.

Sections

These Menu items allow you to take advantage of Samplitude's ability to simultaneously display up to 3 different "views" of the same project.
If you select '2', Samplitude will display the project in two windows. Each section can be handled separately. It is possible to represent the complete project in one section, and a zoomed-in version of a certain Range in the other, for example.
The screenshot above shows a VIP that uses 3 views simultaneously, enabling various ways of looking at the first Object on track 1.

Note: You can return to the default, single view mode at any time by pressing the "Shift+B" keys.

The *Sections > 3* mode is especially useful when adjusting loop points in Wave Projects.
The whole sample can be shown in the upper section, while the section to the lower left displays the beginning of the loop Range, and the section to the lower right, the end of the Range. Use the split Range function for this purpose (key "B").
You can also drag Ranges across the section boundaries (i.e. from view to view). Establish the start point of a Range by clicking, then keep the mouse button pressed, and change over to another section. Samplitude will indicate the size of the Range while dragging. Release the mouse button at the desired location to define the end of the Range.

Fix Vertically

Ranges can be dragged in Samplitude horizontally and vertically. If this option seems too strange to you, you can fix the upper edge of a Range(s) to the maximum value and the lower end to the minimum value. Thus, you get the usual representation in Range marking. However, a trade-off is that you are not able to define the vertical extension of a section by choosing the vertical Range button.
In wave projects, a vertical area selection can be very useful in order to quickly zoom in on the upper edge of the wave form (by means of the "Zoom into Range" button on the position bar). Using the Grid, you can then more precisely read peak values.

Hide Submix/AUX busses

Shows/Hides Busses (if present)
You can access the same function via the "Busses" button, in the top left corner of the VIP window.
Please refer to the "Working with the Mixer" chapter, in the "Quick start/ User's Guide" for more details.

Show Grid

The Grid display is designed to assist you with general timing information, and precise placement of Objects and Ranges. The Grid shows you information about your location (i.e. where you are) in the project window. It can be displayed in several formats, assisting with easier identification, and a better overview, depending on the task being performed.
In addition to identifying locations in the project, the Grid can help you to structure your VIP song, aligned with bars and beats. Precise Object looping is easy when using the Grid.
The Grid display consists of vertical lines that are displayed at specific intervals in the VIP, or Wave Project window. The frequency of the line display is changed by altering the units of measurement. As an example, if the Grid is displayed in Bars & Beats, the vertical lines are very close to each other, indicating where the individual beats occur.
If you zoom into a window, the resolution of the Grid display will also change. As an example, if you wish to see the vertical lines of the Grid every second (i.e. a line on every one second increment), you may need to zoom further into the window.
The *Show Grid* Menu item is a switch that activates/deactivates the display of the predefined Grid in the project window. The units of measurement defined by *View > Units of Measurement* will appear in the upper sections of the Grid. Decibel indicators will also be displayed at the right boundary of each Object.

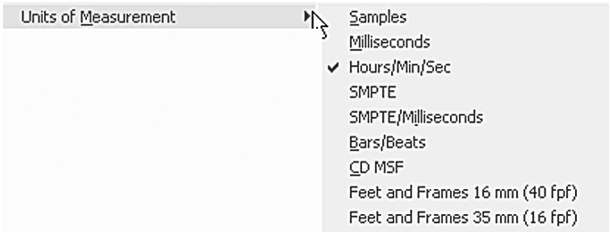
Shortcuts: #

Grid Lines

This provides 6 options for the graphical display of Grid Lines.

Units Of Measurement

Nine options are available for defining the Units used in the Grid. The choice of Unit will depend on the type of project you are working on, although you can easily toggle between the various Unit options in any project.



The units are as follows: 'Samples', 'Milliseconds', 'Hours/Mins/Secs', three SMPTE Frame options, the MSF format for red book CD's (SMPTE at 75 frames), SMPTE + Milliseconds (instead frames) and 'Bars' for BPM display, and feet and frames for 16/36 mm film.

Snap active

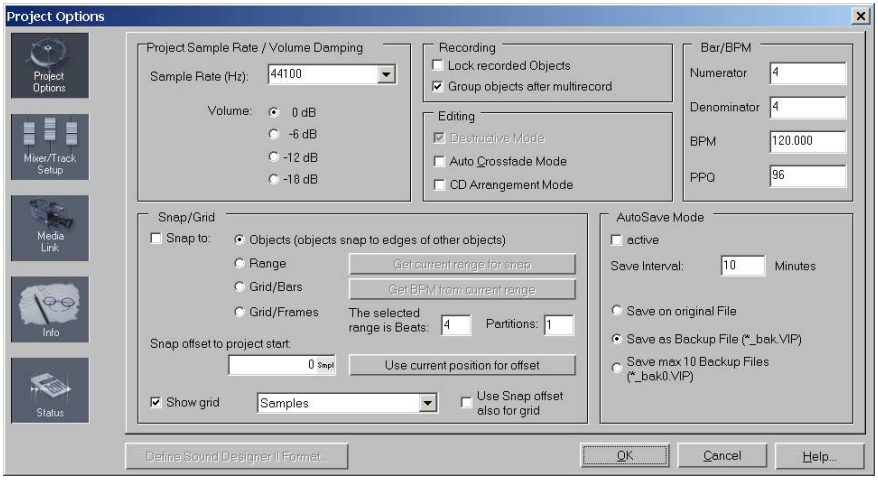
This function switches the Snap on and off. Time positions of Objects and Range borders, changed by dragging with the mouse, are forced to distinct values with Snap active, according to the selected "Snap to..." type.
There is a difference between the Object Snap and the other "Snap to..." types. The Object Snap (the predefined "Snap to..." type) means that Objects and Range borders can be shifted around until they reach the proximity of a reference position. Then they jump exactly to the reference position. The proximity is not a time value but an absolute screen pixels value and therefore independent from the actual project zoom depth. The reference positions are the beginning, the end or the hot spot of all other Objects and markers in the VIP.
If several Objects have been selected, the move is performed by lining up the beginning (or the hot spot) of the last selected Object in the sequence with the reference point. All selected Objects remain in their position relative to each other.
This option can be used to easily rearrange Objects with their audio patterns and gain sample-exact connections.

The other snap types (Snap to Range, Grid/Bars, Grid/Frames) have no proximity rule. If activated, Objects and Range borders always snap to their according positions.

Shortcuts: *Ctrl + ´*

Snap and Grid Setup

This Menu item launches the *project Options* dialog. The dialog provides a number of variables that determine the Snap and Grid settings used in different types of projects.



Snap to: When this box is checked (ticked), the Snap feature is turned on.

Objects: This option activates the Object Snap.

Range : activates the Range Snap and enables you to use the currently marked Range as a Snap basis by selecting Get current Range for snap. This can be used as a simplified form of the Grid/Bar snap. The principle is the same, the Range borders will become snap reference points and these points are continued in both direction in equal distances according to the Range length. Mark a Range with the smallest time fraction (e.g. 1/16 note) and you've got a 1/16 snap. Use this option, when you want to snap to distinct time positions without changing the project tempo.

Grid/Bars: Activates a Snap based on measures. You can enter the tempo in BPM and additional measure attributes in the *Bar/BPM area*. The tempo can also be detected from the current Range selection via "Get BPM from Current Range". You can define the number of beats to which the current area selection corresponds to - e.g. 4, if a complete 4/4 measure has been marked. The program now automatically displays the tempo (bpm) for the marked area.

Partitions are used to define the number of Snap positions per beat, i.e. a value of 4 = 1/16 notes in a 4/4 measure.

Grid / Frames: Activates the snap based on frame length. The exact frame length is dependent from the selected SMPTE format (Options > Synchronization Settings).

Bar/Beat: Here you set the meter (numerator/denominator), the tempo in beats per minute (bpm), and the timer resolution in peaks per quarter (clicks/pulses per quarter note – often shown as PPQN).

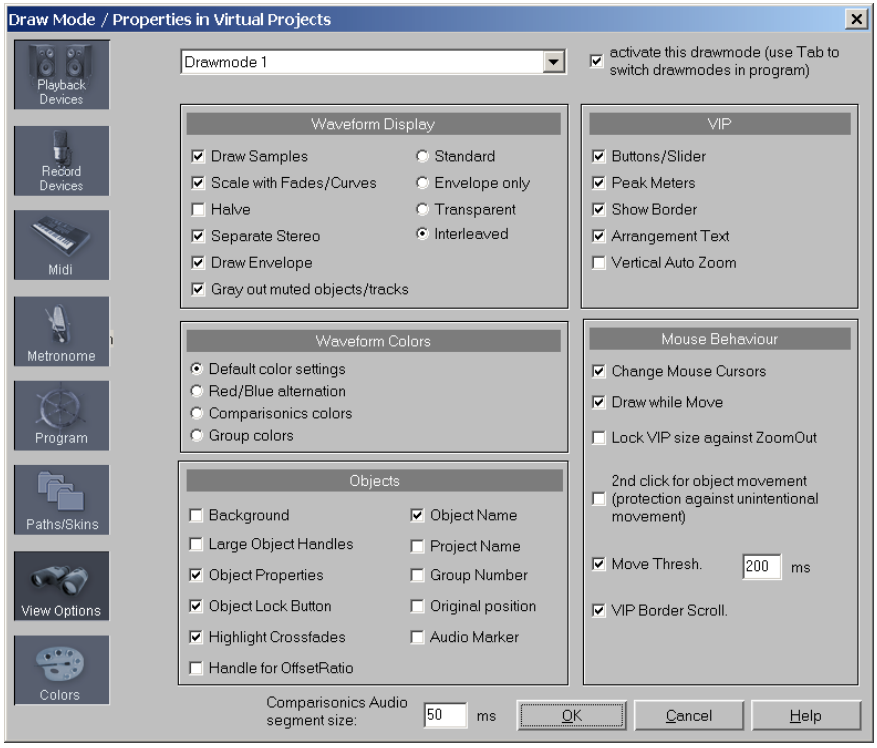
Snap Offset to project start: Lets you define a starting point for the snap and grid.

Use Snap Offset also for Grid: When this box is checked (ticked), the Snap Offset (if set) is applied to the Grid.

Shortcut: *Ctrl+ Shift + ","*

VIP Display Mode

The VIP Display Mode preferences help you to define the two possible alternative display modes, when working with Virtual projects.



You will usually choose mode 1, as it provides detailed overviews of samples (with all information displayed). Mode 2 employs a quick drawing method, without graphics. Perhaps you may prefer using mode 1 to display half mono waveforms, and mode 2 to display full waveforms, in dual stereo. Switching between the two modes is via the Tab key.
You can also toggle between Mode 1 and Mode 2 by selecting the appropriate mode from the Menu item. The Modes are defined by selecting the Draw Mode dialog, which opens the following window:
As shown above, the two Draw Modes have identical options - which are listed side by side for easy comparison. The active mode is set by selecting the appropriate radio button at the bottom of the dialog window.

Waveform Display

Draw Samples: The audio sample representation is switched on. Usually you'll leave this on, but it can be advantageous to deactivate sample representations, when you wish to see volume and pan curves more clearly.

Note: If drawing mode 2 (see above) is activated, the waveform display in the default setting is switched off. You can, of course, change this at any time.

Scale with Fades/Curves: In this default representation mode, the waveform overview is scaled in accordance with the settings of the Fade In/Out, or Volume curves. This makes the decay of the sound during a Fade Out, or Volume Curve changes, more visible. This mode also offers good visibility of the audio material..

Halve: The graphical representation of the audio material can be displayed as a whole waveform, or half, for higher definition. Checking this box will display the waveform with half its information.
This mode is recommended, because the Grid (key #) and vertical Zoom functions (Ctrl + cursor up/down) are perfectly adapted to this draw mode. The display of Volume Curve results on the signal is also improved in this mode.

Separate Stereo: If this option is active, stereo tracks in VIPs are displayed as a two-channel waveform. This allows you to expand the display to both channels, rather than showing a single composite channel.

Draw Envelope: With this option on, an outline is drawn around the envelope of the waveform. In most cases, this provides easier viewing of the waveform.

In the second column you can set the representation of the crossfade.
Standard: The waveform of the second Object is drawn over the waveform of the first one

Envelope only: When this radio button is selected, only the waveform outline (envelope) is drawn. Both Objects are well visible in crossfades.

Transparent: When this radio button is selected, the colors of both crossfade waveforms are overlapped. This works best in front of a dark background (default). We recommend that you combine this, and the Draw Mode option (*Waveform Colors > Red/Blue alternation)*, to optimize the display of crossfaded areas.

Interleaved: When this radio button is selected, a sample of the left Object, and a sample of the right Object, are drawn Alternatively within a crossfade. You can then visually assess the fade area - a task made especially easy if the adjacent Objects are of different colors.

Waveform Colors

Default color settings: All Color Settings are set to their defaults *(Options > Program Preferences > Color setup)*

Red/Blue alternation: This mode alternates the waveform color of adjacent Objects between red and blue. This improves the visibility of crossfades in the transparent and interleaved drawing modes (see above).

Comparisonics colors: This mode uses an algorithm patented by Comparisonics® to depict sonic material in different colors. Dependent on the pitch, and additional parameters, a specific color (hue) is computed and displayed for each segment of audio material.
Low pitches are blue in shade, whereas high pitches are green/yellow/red. Tonal sonic material is clearly shaded in color, while noise and atonal audio is gray in color. This makes it possible to visually recognize the different pitches of a lead saxophone, or to visually detect inconsistencies which would be unrecognizable, if a purely graphic envelope representation was used.
The size of the audio segment, used for the computation of color, can be defined in the Comparisonics Audio Segment Size (see the lower-right of the Draw Mode dialog) field. We recommend sizes of 50 ms or larger, to ensure that sufficient audio material is available for the color computation.
A completely unique audio search method is provided within Wave Projects, based on the Comparisonics colors. This is accessed via the Range > Comparisonics Audio Search Menu Option, and allows you to quickly and conveniently find identical, or similar, audio material in the Wave Project. Please refer to the Menu Items – Range chapter for more details.

Note: For further information on the Comparisonics algorithm, refer to www.comparisonics.com.

Group colors When Objects are grouped, they will automatically be given a new color. So you can quickly identify the Objects groups in the VIP

VIP

Activate/ deactivate Several VIP components (Buttons/Slider, Peak Meter, Border and Arrangement Text) here.

Buttons/Slider: This is the Track Box including all Mixer Control elements like Volume Fader, Mute, Solo.... If you work with the mixer window always open next to the VIP, you can remove all the identical functions. This is also the case for the peak-meter.

Show Border: This option determines if a border is drawn between tracks in a VIP. Switch it off to save space within VIP's (with lots of tracks on small displays – e.g. laptops). Attention: If you deactivate the track borders, you lose also the selected track highlighting.

Arrangement Text: The arrangement text serves to distinguish the two editing areas in Universal mode (area/Object) and is also displayed in this mode.

Objects

Background: Each Object in the VIP track can be assigned it's own color. The color is displayed if this option is checked. If unchecked, the track background color is used. Object color schemes can be assigned via the Object Menu or in the Object Editor.

Object Name: Objects in VIP windows can have their own name. If this option is checked, the Object name is displayed. Object names can be assigned via the Object Menu.

Large Object Handles: When selected, larger handles for Objects and volume / panorama curves will be displayed. This option simply provides better visibility.

Project Name: Samplitude also allows you to display the audio file name in each Object. This is especially useful if you quickly want to identify the Wave Project used in an Object. Checking this option will display the audio file name in the Object.

Object Properties: This shows you which Object-related effects are applied to the Object, e.g. EQ for equalizers, DYN for dynamics etc....

Group Number: Objects in VIP windows can be grouped together. When grouping Objects, it is helpful to have the group number displayed in the Object. This option allows you to display group information in the Object.

Object Lock Button: When locked, an Object cannot be moved. Activation of this option allows you to change the lock/unlock status of an Object, by clicking on the key icon in the lower region of the Object. The key is not displayed if the Object length is too short to allow it's display.

Red key: Object is locked
Yellow key: Object can be moved
Original position: The original recording time position of the Object start in the selected time format.
Highlight Crossfades: This setting applies an alternate Object background color to crossfade to increase its visibility

Mouse Behavior

Change Mouse Cursors: This option causes the mouse pointer to change, when moving it over certain areas. As an example, when moving the mouse pointer over an Object handle, the mouse pointer changes into a directional pointer. This option is useful for mouse modes that combine several functions.

Draw while move: To increase graphic performance, disable this option. When deactivated, only the Object outline is drawn while moving/dragging Objects with the mouse button depressed.

Lock VIP size against Zoom Out: If the VIP is Zoomed out beyond its size (from the start of the first Object to the end of the last Object), it will usually be extended automatically. This behavior can be switched on/off here.

2nd Click for Object movement: This option allows you to move Objects with your second, rather than first, click. The first click simply selects the Object, avoiding accidental replacements/moves.

Move Thresh: You can adjust the Object move threshold value, in milliseconds. This prevents Objects and Markers from being moved accidentally, as soon as you click on them.

VIP Border Scroll: This mode allows you to scroll through the VIP when selecting a Range, and extending it beyond the current window borders.

Shortcut: *Shift + Tab*

Store Position and Zoom Level

This Menu item stores positions and zoom factors. Once stored, they can be recalled with the 8 buttons to the bottom-left of the VIP, the View > Get Position and Zoom Factor" Menu item, or the NUM pad (to the right of the cursor keys). To delete stored positions/zooms, right-click on the desired button(s), or overwrite with a new setting.

The keyboard Shortcuts used to store these settings are shown in the screenshot above.

Note: Remember that right-click = delete setting. Take care, or you may unwittingly delete a favorite setting!

Store Zoom Level

This Menu item stores Zoom factors. They can be recalled with the 4 Zoom buttons to the bottom-left of the VIP, the *View > Get Zoom Facto*r Menu item, or the NUM pad (to the right of the cursor keys). To delete stored zoom settings, right-click on the buttons, or overwrite with a new setting.
The keyboard Shortcuts used to store these settings are shown in the screenshot above.

Note: Remember that right-click = delete setting. Take care, or you may unwittingly delete a favorite setting!

Get Position and Zoom Level

This Menu item loads one of the 4 possible setups for position and zoom factor (assuming that they have been stored via *View > Store Position and Zoom Factor)*. They can also be recalled with the 8 buttons to the bottom- left of the VIP and the NUM pad (to the right of the cursor keys). To delete stored positions/zooms, right-click on the buttons, or overwrite with a new setting.
The keyboard Shortcuts used to store these settings are shown in the screenshot above.

Note: Remember that right-click = delete setting. Take care, or you may unwittingly delete a favorite setting!

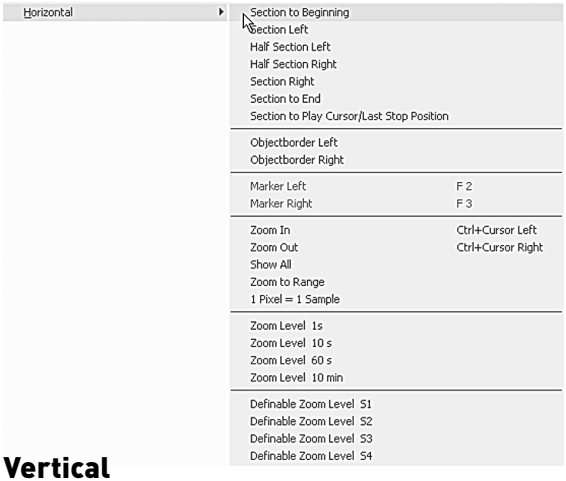
Get Zoom Level

This Menu item loads one of the 4 possible setups for Zoom levels (previously stored via *View > Store Zoom Factor)*. They can also be recalled using the 4 buttons to the bottom-left of the VIP, and the NUM pad (to the right of the cursor keys). To delete stored zoom settings, right-click on the buttons, or overwrite with a new setting.
The keyboard Shortcuts used to store these settings are shown in the screenshot above.

Note: Remember that right-click = delete setting. Take care, or you may unwittingly delete a favorite setting!

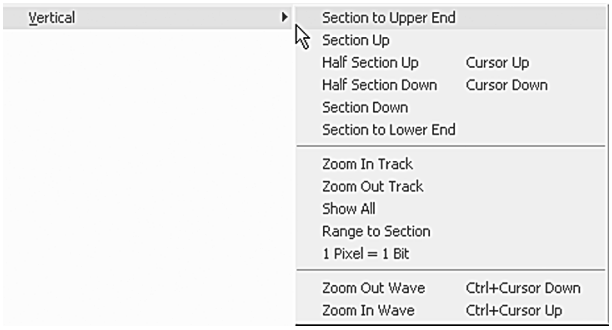
Horizontal

This Menu item contains all functions of the horizontal (red) position toolbar, plus other horizontal view options.



Vertical

This Menu item contains all functions of the vertical (blue) position toolbar, plus other vertical view options.



The various options are self explanatory. We suggest that you experiment with the various vertical viewing options by changing the Menu items, toolbars and Shortcuts. This will help you to establish the options which best suit your working methods.

A little persistence and patience goes a long way – time spent developing your favorite viewing method/system will be repaid tenfold by the speed benefits you will enjoy in future Samplitude projects!

“Track” Menu

This Menu allows you to make various changes to track layouts within a VIP.

Insert new Tracks → Add one Track

Expands the Virtual Project by one track, at the bottom of the VIP track list.

Insert new Tracks → Add several Tracks

The Virtual Project is expanded by multiple empty tracks.
The number of tracks must be entered into a small dialog box before they are appended to the project.

Insert new Tracks → Insert empty track

The Virtual Project is expanded by one empty track, which is inserted below the selected track.

Insert new Tracks → New Submix Bus

The Virtual Project is expanded by one SubMix Bus track, which is inserted below the selected track.
A Submix bus is a track with all capabilities of a normal track, e.g. EQ, FX, plug-ins. The difference is that it allows you to route the Outputs of all lower-numbered tracks into it - i.e. the SubMix Bus. Once a SubMix bus is created, it appears in the output device list of all lower-numbered tracks.
Please refer to the Mixer chapter for details!

Insert new Tracks → New AUX Bus

The Virtual Project is expanded by one AUX Bus track, which is inserted below the selected track.
An Aux bus is a track, which has all capabilities of a normal track, e.g. EQ, FX, plug-ins. The difference is that it allows you to route the Outputs of all lower-numbered tracks into it - i.e. the Aux Bus - by using the aux send sliders in the Mixer. Once an Aux bus is created, it appears in the output device list of all lower-numbered tracks.
You can also send Objects to the Auxes, a great feature for automating Aux effects.
Please refer to the Mixer chapter for details!

Insert new Tracks → New Surround Bus

You can also mix your project as a surround version later. Use this command to create a surround bus. All tracks that are to be 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 can also be routed to this bus and positioned in the surround panorama independent of the track panorama settings. (Object-Surround)

The surround master is created simultaneously when a surround bus is created; the individual channels are routed to the different output devices (more details in the help on Project Surround Setup).

Insert new Tracks → New Surround AUX Bus

This command lets you create a new Surround Bus.
Aux busses are generally used for implementing send effects. The function of a surround AUX bus is the same, and is enhanced by the possibility to include surround effects. It can be fed from a normal track (refer to Stereo and mono signal processing in surround projects) or a lower surround bus (according to the track number). The AUX bus itself can again feed a surround bus of a higher track number.

Cut Track(s)

All tracks in the VIP which are covered by a marked Range are cut and copied into the VirtClip
If no Range is marked, the selected track is cut.

Copy Track(s)

All tracks in the VIP which are covered by a marked Range are copied into the VirtClip
If no Range is marked, the selected track is cut.

Insert Track(s)

All tracks in the VirtClip are inserted into the current VIP above the actual selected track or Range.

Delete Track(s)

All tracks in the VIP which are covered by a marked Range are removed from the VIP
If no Range is marked, the selected track is removed.

Track Properties

This submenu allows access to all track properties (this can also be done by clicking the corresponding button in the track box of the corresponding track in the VIP window). The main purpose of the menu is to remap keyboard shortcuts to Samplitude functions. This allows you to operate Samplitude with the keyboard.
In addition, the menu also allows you to switch the track type, enabling it's use as an Aux or SubMix bus
All menu options affect the selected track (i.e. the one outlined in yellow).

Command	*Description*	*Shortcut*
Mute	Mutes the track	Alt + M
Mute inactive	Mutes, and deactivates the track (this increases performance, because no caching nor FX processing of this track is necessary)	

Solo	Mutes all other tracks, except those that are already "soloed"	Ctrl + Alt + S
Solo exclusive	Mutes all other tracks	Alt+S
Record	Arm track for recording	Alt+R
Record Type	Set the recording type for the track to audio or MIDI recording and for audio recording to Stereo, Mono or Mono mixdown	
Monitoring	switch manual monitoring for the track (See Menu Play/Rec > Monitoring for details)	
Lock	Lock track. No Object moves or length changes are allowed.	
Volume Curve active	Shows / Hides Volume curve	
Pan Curve active	Shows / Hides Panorama curve	
Edit Volume	Open the numeric input field for Volume	
Edit Pan	Open the numeric input field for Panorama	
SubMix Bus	track acts as SubMix Bus	
Aux Bus	track acts as Aux Bus	
Mute Bus Inputs	Mutes the Input of a Bus, the output of the bus is still active	
Global Mute/Global Solo	Enable/Disable all mute/ solo states of the tracks	

Track FX

This option allows (as with the mixer channels) the application of real-time effects to the selected track in Samplitude. More information about effects can be found in the "Effects" chapter.

Track Effects → Track Routing Dialog

Opens the track FX routing dialog .
In many circumstances, you would use the *FX* button in the track Properties box (left of the track), rather than using this Menu item (both perform the same function). Use of the Menu item can be more convenient in some situations, such as when the project track count is high, and the screen view is zoomed out (the *Plug-ins* button is not always visible).
For details regarding the FX routing dialog, read the "Effects in Samplitude" chapter in the manual.

Track Effects → Copy/Paste Track Effects Settings

This command allows you to copy the complete effects settings of the active track into the clipboard (in order to paste them to another track). This includes the order, and parameter adjustments, of all internal effects (EQ, Dynamics, Delay/Reverb...) plus those of (used) plug-ins.
You can also perform this function with the Copy/Paste buttons in the FX routing dialog of the track.

Track Effects → Reset Track Effects settings

Reset all track effects settings to default values - i.e. no effects are applied. You can also perform this function with the "Reset" buttons in the FX routing dialog of the track.

Track FX → Aux Sends

Opens the AUX Sends routing dialog for the track. Please refer to the Mixer chapter of the manual for details!

Track Effects → Track Dynamics/Distortion/Delay/Reverb/EQ/ Plug-ins/Dehisser/FFT Filter/Multi-band Dynamics/Advanced Dynamics/Multi-band Enhancer/Amp Simulation/Room simulation

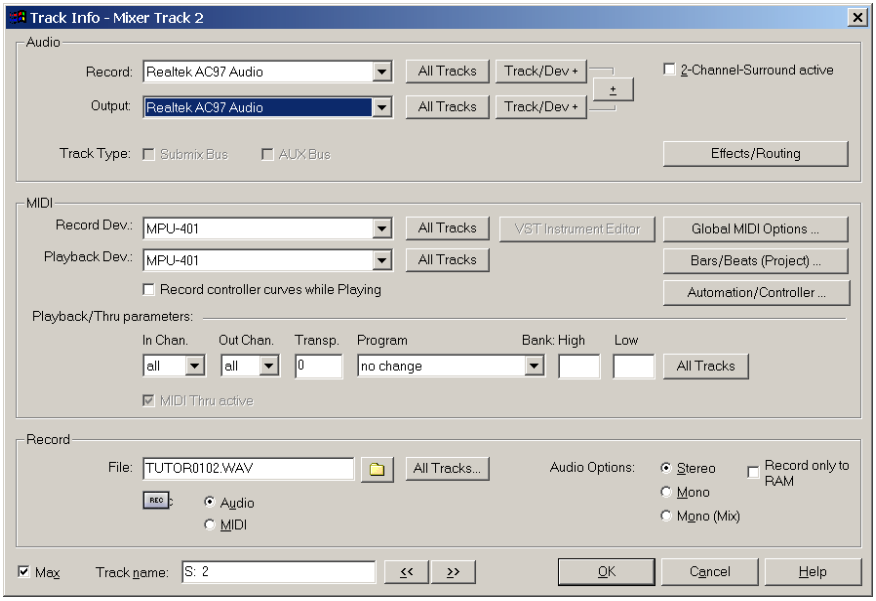
Please refer to the Menu Effects chapter, for further details.

Pan/Surround Editor

The command opens the Surround Panorama dialog. If you are working in normal Stereo Master mode, then the panorama settings for stereo tracks are shown. Further information about the Surround Panorama module can be found in the "Surround" chapter.

Track Information

Opens the track Properties dialog, with the following options:



Audio

Output (Audio Device): A drop-down list selects the audio card (or specific stereo output for cards with multiple outputs) used to play back the track.

Record (Audio Device): A drop-down list selects the audio card (or specific stereo input for cards with multiple inputs) used to record the track.

You can customize the device names listed here as well as disable devices (like the PCs onboard sound system) you never use in the Options > System > Playback or Record devices dialog. Audio devices are organized always as Stereo pairs.

All tracks: This button (just to the right of the two drop-down lists) allows you to quickly assign ALL audio tracks to playback or record through the selected audio card (or stereo output).

Track/Dev +: By clicking on this button, you switch to the next track and assign the next audio playback or recording device to it at the same time. If a track is armed for recording and you click this button for the recording device, this state is also taken over to the next track. The + key between the two keys sets both playback and recording device.

Track Type: Here is indicated if the track is a aux/submix bus. You can change the track type in the Menu track- Track Properties menu item.

2 Channel Surround on: Check this box to apply a 2-Channel-Surround Sound to the track. Read more about this in the Surround chapter

Effects/Routing: Opens the FX routing dialog for the FX-Routing settings for the track

MIDI

Playback Dev: Selects the MIDI playback device for the track.

Record Dev.: Selects the MIDI record device for the track.

All Tracks: Sets the MIDI record / play device for all tracks.

Global MIDI Options: Opens the MIDI Options dialog.

Bar /Beats (project): Opens the Bar Definitions dialog.

Automation/Controller: Opens the MIDI Controller / VST automation Curves Settings dialog.

Record Controller Curves: Enables recording of MIDI controller curves from MIDI devices.

Playback/Thru Parameters: The data received at MIDI IN is output via MIDI OUT. MIDI events are received on all or on a specific channel, which you control by the In Chan. Setting. The events can be transmitted on their original channel or on a different channel (set by Out Chan.). In addition a Transpose

(Transp.), Program Change and a Bank Switch command can be transmitted. Transpose affects all MIDI Objects in the track.

MIDI Thru active: Enables MIDI through for the track.

Record

Record File: You can enter the name for the audio file you are about to record, in this field. Click on the yellow folder button to open the standard Windows file naming / browser dialog. Normally these files are named after the VIP name with a suffix number for the track number and created in the VIP subdirectory. When you edit your whole recording project in the VIP as it is intended by the concept of Samplitude's virtual working, there's no need to change something here. Nevertheless you can: by clicking on the folder button you may enter a new name for the audio file or select a new location.

Record to RAM: Check (enable) this option to record to your system's RAM (RAP recording) rather than the hard drive (HDP recording). You should only use this option for very short recordings! Be careful not to overtax your system's RAM, or the system will become sluggish and unstable.

All tracks: With this button you can name/rename quickly the recording files for all tracks. Several different naming conventions are possible to construct the name from the track number, track name, project name or the given file name from the file name edit field.

Stereo: Click on (enable) this option to record in from a stereo input device into a stereo Wave Project.

Mono: Click on (enable) this option to record into a mono Wave Project. If you select this option, the audio recording device selector changes to mono devices, actually the left or right channel of a stereo input pair. Make sure you select the right one (Left or Right channel)

Mono (Mixdown): Click on (enable) this option to record from a stereo input device into a mono Wave Project by adding the contents of both channels and applying a -6 dB attenuation.

Record Mode: These two radio buttons allow you to select either audio or MIDI recording. Although you can use a track for a mixed audio/MIDI playback you cannot record both type of data at the same time on one track.

Track Name

You can type a name for the track here. When you open the Mixer window, this name will be used to identify the track's mixer channel. Use the << / >> buttons to rapidly switch between the tracks, allowing you to set up all your track settings with no need to close and reopen this dialog.

MIDI Controllers/VST Automation/VST Instrument Editor

Please refer to the "Working with MIDI in Samplitude" chapter.

Track Visualization

The visualization from the menu window can be set as Master display or track display. For further information about different views please refer to the "Menu Window" chapter.

Hide Track

Hides the selected track from the VIP display. It continues to be visible in the mixer and is also played back.
An overview of all tracks including their status of visibility is included in the track Manager. ("Tools" menu > Manager > Track Manager or "Ctrl+Shift+S" keyboard shortcut). This also lets you hide a track in the mixer window.

Minimize Track / Maximize Track / Minimize none

Minimize track: This Option minimizes the selected track. This can also be done with the + button in the Track Box in the VIP Window

Maximize Track: This Option minimizes all tracks except the selected track. The selected track will be zoomed vertical to the maximum possible level

Minimize none: This Option un-minimizes all tracks.

Track Freeze/Edit Track Freeze/Track Unfreeze

Please refer to the *"Freeze"* chapter for further details!

Activate next

Moves the track selection up/down.
Alternatively, you can use the Alt+ cursor up/down keys.

Activate previous

Moves the track selection up/down.
Alternatively, you can use the Alt+ cursor up/down keys.

"Object" Menu

Contrary to the menu Edit all functions in this menu exclusively manipulate selected Objects in Virtual Projects. Ranges are used to define the insertion points, or cut positions, of Objects. When inserting material into a Virtual Project, the Range start point performs much the same function as the Play Cursor – i.e. the material is inserted at the Range start point.
Splitting, and splicing, are two of the most commonly performed Object editing tasks. The beauty of Samplitude, for these duties, lies in the fact that Objects are virtual – i.e. no changes are actually made to the physical audio files. This allows for very fast editing and flexibility.
The Object Editor is also a fundamental Samplitude feature, providing powerful, and individual processing for each Object.

New Object

Use this command to create a new virtual Object. Before doing so, you must first select a Range in the Wave Project that you wish to reference. The current Range in the Wave Project will be inserted into the Virtual Project, at the Play Cursor position. It will automatically become the currently selected Object. The results of this procedure are the same as the Drag & Drop method of importing Wave projects into VIP tracks.

Note: The Object will be inserted at the actual Range position in the VIP. If a Range is not marked in the VIP, the Object will be inserted in track 1, at the current Play Cursor position.

Cut Objects → Cut Objects

The currently selected Object(s) is replaced with empty space, and copied to the VirtClip. The length of the current project is unchanged, and all non-selected Objects remain in their original positions. The previous contents of the VirtClip are replaced with the Cut Object.

Cut Objects → Copy Objects

Select this option to place a copy of the (currently selected) Object(s) into the VirtClip. The contents of the VirtClip can then be inserted into the project with the *Cutting Objects > Insert Objects* option (see below). The previous contents of the VirtClip are replaced.
Note: You can also Copy/Paste Objects by click-dragging, with the Ctrl key depressed.

Cut Objects → Insert Objects

Objects in the VirtClip can be inserted at the current Play Cursor position of the project. The other Objects in the project maintain their positions. Please note that the newly-inserted Object may cover portions of existing Objects. Simply drag the new Object to a different set of tracks, or reposition the Object on the same tracks.

Cut Objects → Delete Object

The selected Objects are deleted from the current project. The length of the current project, however, remains the same. Unselected Objects maintain their positions. The contents of the VirtClip remain unchanged.

Cut Objects → Extract Object

Effectively the inverse of *Delete Object*. All unselected Objects are deleted. Please note that this function is similar to the *Edit > Extract* Menu item. Where the two differ is that the Edit Menu function requires that a Range be selected, prior to it's use.

Cut Objects → Duplicate Objects (in place)

All selected Objects are duplicated, and pasted to the same position as the original Objects. The original Objects are overlaid with the copied Objects. The duplicated Objects can easily be shifted to any desired position with the mouse. If several Objects are selected before duplication, press the shift key before moving them, to ensure that all Objects remain selected, and are moved together.
This option does not make use of the VirtClip, so any existing VirtClip contents remain unchanged.
This function is also available through use of your mouse. Press-hold the Ctrl-key, click on the Object you wish to duplicate, and drag the Copy to a new position in the project.
Another option is to use the Drag & Drop functionality for the Object, or the Universal Mouse Mode:
If multiple Objects are selected, press the Ctrl key and drag the selected Objects to the new position. This allows you to duplicate the selected Objects easily, without using the VirtClip.

Cut Objects → Duplicate Objects

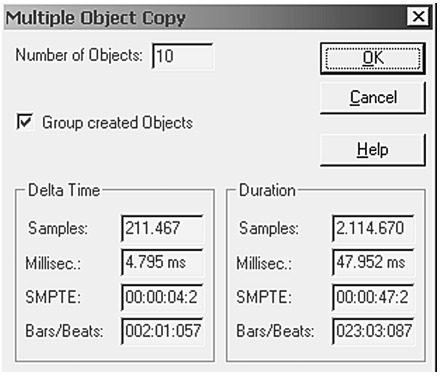
All selected Objects are duplicated, i.e. a copy is created and placed behind the old Object (depending on the grid).
This command is also available in the MIDI Editor.
Duplication is also possible via drag & drop (in Object or in universal mode): If several Objects have been selected before duplication, the Ctrl key should be pressed before subsequent moving so that all Objects remain selected and are moved together.
This way, Objects can be reproduced directly via VirtClip.

Shortcut: *Ctrl+D*

Cut Objects → Duplicate Objects multiple....

This function duplicates multiple Objects, and places them in sequence. Another dialog is displayed (see the following screenshot), which allows you to determine; the number of copies (of the selected Objects), the distance between

the duplicated Objects, and the overall length of the duplication. This is an excellent method for quickly building loop-based projects.



Number of Objects: Determines the number of Objects that are created.

Group created Objects: When checked, all created Objects are grouped together.

Delta Time: This determines the relative position of each created Object, to the next (from the beginning of one Object to the beginning of the next). The default setting is the length of the selected Object. At the default setting, the duplicated Objects are adjacent to one another, creating a seamless Loop.

Duration: As an alternative, the duration can be specified, thus combining the time difference, and the number of duplicated Objects.

Cut Objects → Split Object On Marker Positions

The Object is split into several Objects at all VIP Marker positions that lie within the Object borders.
The new Objects have the same Object properties (FX, Object volume, color...all set in the Object Editor).

Cut Objects → Split Object On Track Marker Position

The Object is split into several Objects at all CD Track Marker positions that lie within the Object borders.
The new Objects have the same Object properties (FX, Object volume, color...all set in the Object Editor).

Split Objects

This function splits or separates selected Objects so that individual Object sections can be further manipulated. All selected Objects are separated at the current play cursor position, which results in two independent Objects.
If a Range is selected, the cut occurs on both Range borders. It is not necessary to select Objects prior to choosing the Range and calling the Split Objects

function. All Objects affected by the Range are split in this case. If an underlying Object is separated, the newly created Object covers the original underlying Object.
The two new Objects have the same Object properties (FX, Object volume, color...all set in the Object Editor).
If the command is performed while playback, the split position is defined by the actual play cursor position at the moment you perform the command.

Shortcut: *T*

Trim Objects

This function trims the borders of the selected Object(s) to the currently selected Range borders. The selected Range needs to be located inside the borders of the Object you want to trim.

Shortcut: *Ctrl + T*

Heal/Unsplit Objects

With this command you can join Objects which were split into pieces (e.g. by accident). The Objects have to border each other in the VIP as well as its referred audio data has to fit. That means, the have to refer to distinct, but bordering wave data of the same Wave Project. If this conditions are not fulfilled, the command is grayed out. All Object settings (fades, effects etc.) of the second Object are removed, the resulting Object carries the settings from the first one.

Glue Objects

The option discussed in this section is related to the freeze function. The new audio data is copied to the freeze directory and remains there for as long as the grouping state is maintained.
The "Group Objects..." command—found in the Object menu and Object context menu—leads to a destructive bouncing operation, whereby a new Object is created through combining several individual Objects, including effects, time-stretching settings and crossfades. The new Object has no effects. Fades remain unchanged. Object aux sends of the first Object are applied to the new Object. By calling the Unfreeze function, the original state of the Objects is re-established.
By using the "Edit Object Freeze" command, grouped Objects can be recalled and edited as a Freeze Project in the state they were in when the grouping command was executed. All changes performed here are automatically applied to the grouped Objects.

Object Freeze

This function renders each individual Object to a new audio file which then replaces the original Object in the VIP. This is advantageous if, for example, you want to activate very system-draining real-time plug-ins. The original Object

always remains preserved and can be re-edited using the "Edit Object Freeze" or can be recalled using "Object Unfreeze".
If more than one Object is selected, the function will be applied to each individual Object. Fade-in, Fade-out and Object volume are not calculated, as these properties are taken over by every new Object.

Note: To make a single Object out of a many Objects, select the appropriate Objects in the track and use the "Glue Objects" function.

Object Freeze → Edit Object Freeze

Opens the root VIP of the frozen Object. This VIP contains tracks with the original Object(s).
In the case of "Glue Objects" the root VIP contains more than one Object, whereas for "Object Freeze" it contains only one.

Warning: The length of the root VIP can not be changed, as the length is set by the Object to which the "Freeze" or "Glue" function was applied.

Object Freeze → Object Unfreeze

The function can only be used on Objects which were created by "Glue Objects" or "Freeze Objects".
It recalls the Object(s) stored in the root VIP. Changes made in the root VIP will be included.

Tip: If tracks are added to the root VIP, it is not possible to use "Unfreeze Object". The root VIP is the project which was created by freezing a track or Object.

Lock Objects → Lock Objects

To protect Objects from unintended editing use this option. First, select the Objects you want to "lock" and activate the lock function. Locked Objects are marked by turning the little key symbol red.
You can define which the forbidden editing actions (moving, duration change ...) in the Object Lock definitions dialog (see below).
You can also lock/unlock a single Object by clicking its lock symbol directly. Locking a track means activate Object Lock for all Objects on this track.

Lock Objects → Unlock Objects

Any selected (and locked) Objects will be unlocked

Lock Objects → Lock Definitions

Selects options for Object locking (as shown in the screenshot). Only "moving" is locked, by default. Locking a track means activate Object lock for all Objects on this track.

Disable Moving: Disables moves of selected Objects (default). This is useful in multi-track projects, when performing quick editing.

Disable Volume Changes: The volume handles are disabled for the selected Object/s.

Disable Fade-In/Out: The fade handles are disabled for the selected Object/s.

Disable Length Changes: The length handles are disabled for the selected Object/s.

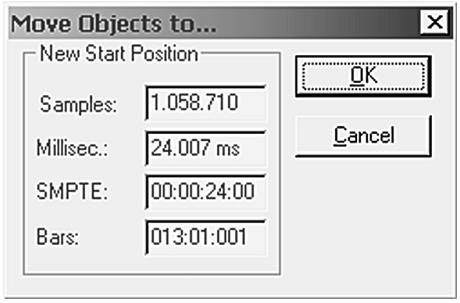
Disable Deleting: This option prevents deletions of locked Objects. This option only affects the deletion of selected Objects by pressing "Del" (or the "Delete Object(s) from the Object > Cutting Objects menu). It will *not* prevent the deletion of an Object when the Range that encompasses the Object is deleted.

Move/Edit Objects

This Menu branch contains a wide variety of options for fine-tuning Object position, especially when crossfading.

Move/Edit Objects → Move Objects

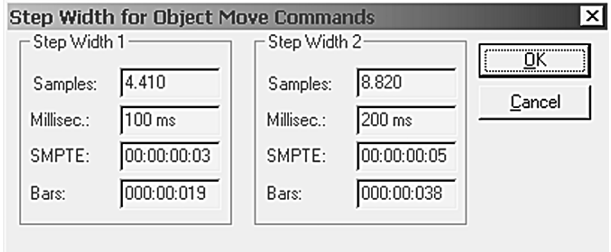
This function allows you to move an Object to a precisely specified position.



The dialog shown above is opened, enabling you to select the new position in samples, milliseconds, or SMPTE time.

Move/Edit Objects → Object/Fade Step Settings

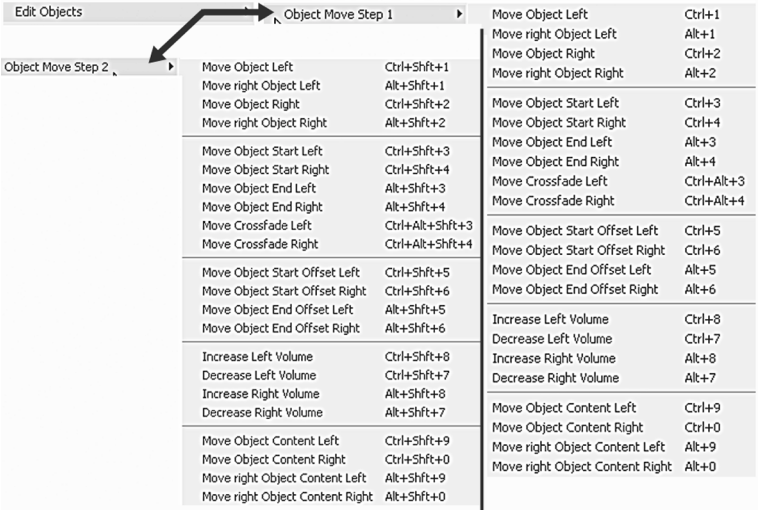
Samplitude contains a complete function set for editing Object fades and crossfades. To edit the crossfade, select two overlapping Objects. To edit a single Object, select only one Object.



All of these functions can be adjusted incrementally, allowing for very precise Object movements. The settings can be defined via the Object->Object/Fade Step Setting menu.

Move/Edit Objects → Object Move Step 1 and Step 2

These Menu items are primarily used for fine-tuning crossfades. You can move all handles of one or two selected Objects, with these menu functions.



Please note that although these functions can be executed from the Menu item, it is far quicker to use the associated keyboard commands to execute the functions.

In combination with the "Activate next/previous track" (Alt +cursor up/down) and the "Select next/previous Object" (ctrl+alt+cursor left/right) allows you to complete project edits, without using the mouse at all!
The basic principle of manipulating the Object is as follows: The Ctrl key selects the left Object, the Alt key the right Object. The increment length (Step length 1 or 2) is determined with the Shift key.
The number keys (0-9) select one of the following functions for the selected Object:
1...move Object left
2...move Object right
Quite self-explanatory

3...move Object start/end left
4...move Object start/end right
With two selected Objects, the ctrl and alt keys select either; the end of the first, or start of the second Object. Ctrl + Alt moves the whole crossfade. When only one Object is selected, the modifier keys select the start and end points of this Object.

5...extend fade time
6...reduce (shorten) fade time
As above: With two Objects selected, it affects the fade out of the first, or fade in of the second Object. As long as you don't have asymmetric crossfades enabled (please see the Crossfade Editor chapter), you'll notice that Alt + 5/6 have no effect, because the fade in/out times of the crossfade are linked. When only one Object is selected, the modifier keys refer to the fade in and out points of the Object.

7...increase Object volume
8...reduce Object volume
This is the equivalent of moving the Object volume handles up and down.

9...move Object content left
0...move Object content right
This is a very special facility, that can only be accessed in this way or via the Object Editor. You can move the Object's underlying audio material. This is the equivalent of: moving the Object's start point, moving the whole Object in the opposite direction, and the Object's endpoint by exactly the same amount. To illustrate, just open the Wave Project window of an Object (Wave Editing), and use the keys. You'll see the effect as you move the Range in the Wave Project, which addresses the audio data for the Object.

To perform all of these manipulations with a Step length of 2, simply use the Shift key, prior to selecting the numeric value (i.e. Alt + Shift + 0 = move content of the second Object right by Step length 2).

Move/Edit Objects → Object To Play Cursor Position

Moves the selected Object to the current play cursor position. If multiple Objects are selected, the selected Object with the smallest track number is moved to the play cursor, all other selected Objects are moved relative to it (in consideration of the Object modes).

Move/Edit Objects → Object To Original Time Position

This command moves the selected Object to the original time position of the material's recording/creation. This can be helpful if an Object is accidentally moved, and you don't notice the change until a few edits later, or even a later session.

Note: The command only reverts to the time position – if an Object has been moved to another track, it will be placed at it's original location on that track, not at the location of the track it was originally recorded/created on.
If an Object has been moved between tracks, the only current method of reverting to earlier position/s on other tracks is via Undo, or the use of VIP backups!

Move/Edit Objects → Set Original Time Position

This Menu item sets a new time position for the 'Original Position' attribute of the selected Object.

Note: The command only sets a new original time position – if an Object is subsequently moved to another track, it will be placed at the new time position on that track, not at the location of the track it was originally recorded / created on.

Move/Edit Objects → Arrange Objects

You can arrange all selected Objects in a way that each Object is separated from the next by a given distance (time). This is useful when you prepare a CD master.
You can also select Objects across several tracks, the arrangement of the Objects will happen individually for each track. The first-selected Object in each track will always remain in its original position.

Mute Objects

Playback of the selected Object/s will be muted – i.e. silent. The waveform will remain visible.
The letters MuL + MuR will appear in muted Objects, indicating that both left and right channel is muted. You can mute left and right separately in the Object Editor.

Shortcut: *Ctrl+M*

Build Loop Object

With this function, a Loop can be defined within an Object. A Range, which indicates the Loop length, must first be selected in the Object. If there is no Range, the whole Object is looped (Loop start/end = Object start/end)
The Object then turns to a "Loop Object". This means that the number of Loops can simply be raised with the 'Object length handles.
To put it simply, drag a Range over the section to be Looped, select Object > *Build Loop Object*, then click and drag the right edge of the Object to the right. The Loop section will be repeated.
You can edit the loop start/length/stop times, and disable the loop "attribute" of an Object in the Object Editor - on the pitchshift/timestretch tab.
Loop Objects are ideal for generating long drum sequences from a single drum loop! Loop Objects also help to save memory, as only one Object is handled rather than multiple Objects, or copies of the same material or very long samples!
If you split a Loop Object at any given point, and disable the loop attribute on the (new) second Object, the second Object plays back seamlessly from the loop end point in the audio. This allows you to loop a drum hit, creating a Drum Roll, for example. Just select the Range over the hit you want to make "roll", pull the length handle, and split the Object after the last drum hit you

want to hear. Disable the loop flag on the second Object, and the last hit will continue with its natural reverb tail.

Set Hotspot

The current Play Cursor position is turned into a reference point for the Snap function. Once defined, the hotspot is used for snapping the Object to the grid, rather than using the Object's start point.
Dashed vertical lines illustrate hot spots. Hot spots may be positioned outside the boundaries of an Object (to the left or right of an Object).
This function is very useful where you wish to synchronize a portion of an Object, which is not aligned with it's beginning.

Select Objects → Select Objects

This Menu item will select all Objects located partially, or entirely, in the selected Range, or Objects beneath the current Play Cursor position (if no Range is selected).

Select Objects → Select All Objects

All Objects of the project will be selected.

Select Objects → Switch Selection

This Menu item performs the reverse of the command above. All selected Objects are deselected, and all unselected Objects are selected.

Select Objects → Deselect Objects

No Object in the project will be selected. This is useful when you wish to delete a Range with the "DEL" key, because the key gives priority to selected Objects, over selected Ranges.

Select Objects → Select next Object

Selects the next Object on the selected track. If no Object is selected, the command selects the Object nearest to the play cursor.

Shortcut: *">", CTRL + ALT + W*

This command, in conjunction with "Select next/previous track" (cursor up/down), allows you to navigate through the Objects by keyboard only!

Select Objects → Select previous Object

Selects the previous Object on the selected track. If no Object is selected, the command selects the Object nearest to the play cursor.

Shortcut: *"<", CTRL + ALT +Q*

This command, in conjunction with "Select next/previous track" (cursor up/down), allows you to navigate through the Objects by keyboard only!

Select Objects → Object-Lasso

The normal way to select several Objects at once would be to click on the free space on the right of the Object in the lower area of the track and to drag a frame around the selected Objects.
Sometimes, however, there is not enough space between the Objects. The Object Lasso solves this problem.
Once activated, you can click on an Object and pull up a selection frame, without moving the Object you clicked on (which would be the case in the standard mode).
Once you selected your Objects the mouse returns to the standard mode, that means that you will have to reactivate the Object lasso each and every time you want to use it.

Shortcut: Ctrl + Alt + L

Group Objects

Multiple selected Objects are grouped together. All operations are applied to the whole group.

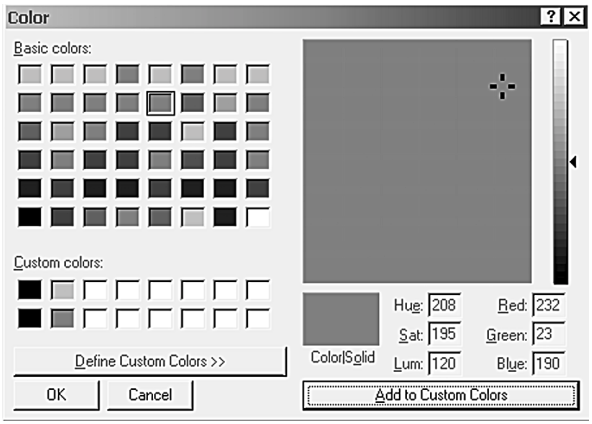
Shortcut: Ctrl + G

Ungroup Objects

Selected Objects are ungrouped. Individual Objects can now be processed independently.

Shortcut: Ctrl + U

Object Color/Name → Object Background Color



Samplitude allows you to specify different colors for selected Objects. The dialog shown above allows you to apply virtually any color to Objects, dependent on your computer's display settings.

Object Color/Name → Object Foreground Color

Samplitude will also allow you to apply virtually any color to selected Object foregrounds. The dialog is identical to the previous dialog (see above).

Object Color/Name → Object Name

Another way to distinguish certain Objects is to give them a different name. Samplitude allows you to specify a name (such as 'Svennevig Shuffle') with this option.
Please note that the name will be only be visible if you enabled the *View > VIP Display Mode > Definition* Menu item, or by pressing *Shift+Tab* when selecting the Object.

Shortcut: *Ctrl + N*

Object Editor

This Menu item launches Samplitude's Object Editor window.
The Object Editor is covered in detail in the Object *Editor* chapter. Please refer to this chapter for further information.

Object Manager

See "Tools" Menu

Take Manager

See "Tools" Menu

Wave Editing

Samplitude is primarily a non-linear, non-destructive multi-track audio editor. This means that most of your work is usually completed in the virtual domain (VIP), without making changes to the original audio. There are, however, some jobs which are best completed by working directly on a Wave Project, as if you were using a conventional audio editor.
Samplitude can also function as a conventional audio editor, but now offers two alternatives, namely: Destructive Editing, or Wave Editing. Samplitude's development team designed a way to circumvent the disk space, and time inefficiencies associated with Destructive Editing/Wave Editing. When working directly on Wave Projects, the powerful options offered by the Wave Editing system place Samplitude a "cut" above other conventional audio editors!
Read the Virtual Wave Editing section for an introduction to the new Wave Editing mode, and also take a look at the Glossary for a rundown on Wave Editing basics.
There are some differences between the "Destructive" and "Wave Editing" modes, when working on Wave Projects:

Rendering edit operations on saving
All edits: cut, paste, delete and insert, are virtual. The edit positions are marked by dotted lines. There is no undo file creation. All edit operations are RAM resident and applied to the Wave Project when you save it.

Master Section: You can use the Mixer, while working in a Wave Project. This allows you to apply real-time FX manipulations as you would in the normal VIP Master section. These effects will be applied to the Wave Project when you save it.
For more information on the Mixer Master section, read the "Working with the Mixer" chapter, in the "Quick start/User 's Guide"!

Menu File Save in format: This will open the track bouncing dialog (format conversions are possible). (Tools > Track bouncing)

Auto Crossfade: The Auto Crossfade option (Edit Menu) works in Wave Projects, too. Default crossfades are applied to every cut or insert operation, when auto crossfade is active. You can use the Crossfade Editor to fine-tune/ edit crossfades.

New File with Drag and Drop: You can mark a Range in the Wave Project, and drag this Range to an empty space on the program desktop. A new Wave Project (untitled.wav) will appear. This method allows you to easily make copies of the entire (or parts of the) Wave Project.

Edit a copy of Wave Content

This opens the Wave Project referred by an Object for Wave Editing. But in unlike the normal "Wave Editing" command it copies the wave data of the Object at the end of the Wave Project and adjusts the Object in a way, that it points on these audio data.
This has the advantage, that the operation creates an undo step in the VIP, so you can undo the Wave Editing without opening the Wave Project again. The disadvantage is, that you change the length of the Wave Project, which is not desired in all cases (when using the audio data in a loop, for instance).

Edit Root VIP

The root project is a special option closely related to the Freeze option. It can be used, for example, for editing bounced Objects in their original state for a mastering project.
Every Object created in Samplitude through bouncing can be edited in this primary project in its pre-bouncing state, as long as its original audio data and primary project (=root VIP) are still available.
This option, in which, basically, a complete project with all its options can hide behind any Object, enables complex audio productions and a convenient recall of the previous working state prior to the bouncing operation.

Operation

This opens a copy of the root project in the state it was last saved in. If it was recalled and edited after the bouncing procedure it may, of course, differ from the previously created audio file.

After editing the root VIP and applying the "save project" function, you are requested to confirm whether the Object that was originally created is to be updated by means of a track-bouncing procedure. All track-bouncing options (format, dithering settings etc.) do apply for this operation. If no other section or a different section exists, the original length of the audio file is valid for this process.

However, the automatic update request only ensues if the original VIP is also called up with the "Edit root VIP" command. A project called up in the usual way using "Open project" has no effect.

”Real-time Effects” Menu

This menu entry permits fast access to the relevant real-time effects at Object level. Correspondingly, menu entries are provided here which can be controlled via the Object Editor.
For more details on these effects please read the relevant articles in the chapter entitled “’Offline Effects’ Menu”.

Object Switch Channels

With this function you can switch left and right stereo channel. This is useful to correct recordings with switched channels.

Object Multi-band Enhancer

The Multi-Band Stereo Enhancer allows you to perform detailed modifications and corrections to the stereo image in three independent frequency bands. The signal is separated into the three frequency bands and the separate bands are processed individually.
The assembling of the independent bands is 100% frequency neutral thanks to the FIR complement Filter technique – no discoloration of the frequency response is introduced.
For More Information about the Effects parameters refer to the “Offline Effects” Menu

Invert Phase

The sample data within the selected Range is inverted along the amplitude axis. This phase inversion means that negative values become positive and vice versa.
The “Invert” function permits samples with different phases to be matched.
Along with the available mixing functions (which are, from a mathematical viewpoint, adding functions) you can actually subtract samples by applying this function to the selected sample.

Object EQ

This dialog contains a 4-band parametric equalizer. You can activate filters on four freely selectable frequency ranges to adjust the sound of a sample. You can produce wide-band frequency adjustments for both high and low pass ranges as well as small-band corrections of specific frequency ranges.
For More Information about the Effects parameters refer to the “Offline Effects” Menu

Object FFT Filter

The FFT filter for real-time application in the Object Editor, in the track and/ or stereo-mix master allows the precise linear phrasing filtration of signals and is thus very suitable for mastering. It contains 1024 frequency-bands, each of which can be raised by up to +30 dB and reduced by around –80 dB.
The FFT filter for real-time application is a “minor” option with a huge number

of possible setting, analyses and function modes for destructive editing of the audio material ("Offline Effects" menu). The filter-curve can be edited by plotting a curve with the mouse.

Object Dynamics / Object Advanced Dynamics /Object Multi-band Dynamics

Opens the corresponding dynamics dialog for the Object. For More Information about the Effects parameters refer to the "Offline Effects" Menu and the "Effects Chapter" in the manual.

Object Dehisser

The Dehisser complements the Noise Reduction DSP. It can be used to remove constant low-level White Noise. This is typically found in material recorded through MIC pre-amps or analog sound cards.
The Dehisser does not need a noise-print sample, as is necessary with the Noise Reduction DSP.
For More Information about the Effects parameters refer to the "Offline Effects" Menu

Object Distortion

This effect allows you to distort audio material by using a non-linear characteristics line; the signal becomes louder and overtones are added. By adjusting the starting point of the distortion (Threshold), a soft, analog sounding distortion (i.e. Threshold at –40 dB) can be created (Overdrive). A harsh, digital sounding distortion can be created with a Threshold setting of 0 dB. The Ratio setting determines the strength of the distortion.

Object Amp Simulation

The Amp Simulation is a real-time effect that simulates tube amp sound properties. It is especially suitable for guitar sounds, but also for the organ or recordings of vocals.
For More Information about the Effects parameters refer to the "Offline Effects" Menu

Object Vocoder

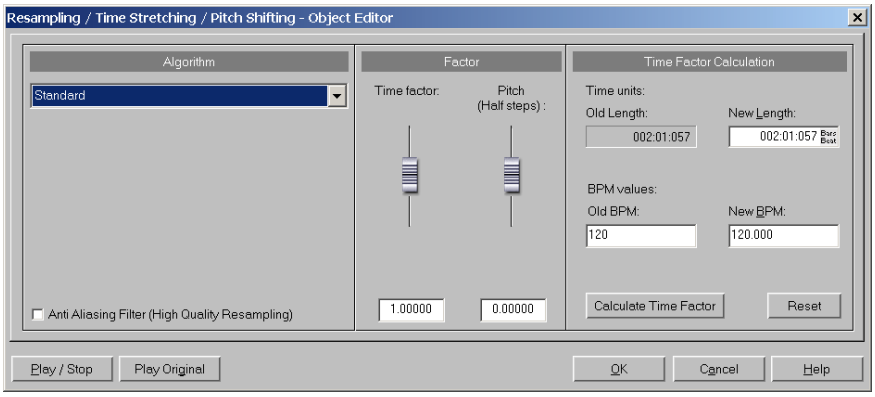
The Vocoder in brief: A carrier material (e.g. a string pad or a Synth chord) are affected by a modulator (e.g. language or singing, or even drum loops) to give the impression that the audio material is "speaking" or "singing".
For More Information about the Effects parameters refer to the "Offline Effects" Menu

Object Room Simulator

This command opens the Room Simulator for the Object. Please make sure, that the Object volume is placed before the Room Simulator effect to avoid fade out of the reverb tail. For More Information about the Effects parameters refer to the "Offline Effects" Menu

Object Resampling/Timestretching

This command opens the dialog for an Object’s assigned resampling or pichshifting algorithm. For example:



For More Information about the Effects parameters refer to the “Offline Effects” Menu and the Chapter “Elastic Audio“.

Elastic Audio

Opens the Elastic Audio Editor. There is a detailed description of the Editor in the first Manual part in the Chapter of same name.

Shortcut: *Ctrl + Shift + E*

Object DirectX / VST Plug-Ins

This function allows you to use Microsoft DirectX and VST 2.0 compatible plug-ins and with Samplitude. This complements the already existing effects with an unlimited number of 3rd-party effects.
For More Information about the Effects parameters refer to the “Offline Effects” Menu

”Offline Effects” Menu

The options available in the Effects menu are processed directly to Hard Disk Projects (HDP), RAM Projects (RAP), or selected Objects in the VIP window. With the exception of the Normalize Object (Virtual) option, the effects are processed destructively. In other words, the original audio file will be overwritten with the processed file. If you enable the *Create Copy* option (in each Effect's window), the processed audio will be appended to the end of the original (dry) audio file.
When processing selected Objects in the VIP window, having this copy allows you to undo the process, and restore the original (dry) Object. The appended audio is retained in the project (even though it may no longer be referenced by the VIP) unless deliberately deleted, either manually or with the *Tools > Remove Unused Samples* function.

Note: If you process a selected Object which appears elsewhere in the project (e.g. a loop-based project) and you do NOT have the Create Copy option checked, you will affect ALL copies of that Object in the project, as the original HD Wave file has been destructively processed. This will not occur if Create Copy is checked, as the processed Object will now reference the appended audio section (see previous paragraph).

We recommend that the *Create Copy* option is always checked when applying effects via the Effects menu. Even though this results in the project requiring additional hard disk space, you can always be certain that you have an undo (or the original file) if you need it. You can never be “too” sure that you no longer need the original file!
If the *Create Copy* function is not checked, you will not have an Undo for the processing, regardless of whether undos for HD Wave Projects are enabled in the *Options > Program Preferences > Undo Definitions dialog*. Your only option will be to restore a backup of the entire project. (We don't need to remind you that regular backup of all referenced files is essential.) This applies to VIP and HD Wave Projects.
Dialogs in the Effects menu that don't offer a Create Copy function (such as the Fade In/Out dialog), will automatically create an Undo file, provided that the Undo is enabled for HD Wave Projects (Options > Program Preferences > Undo Definitions dialog). These Undo files are temporary, and are deleted when the project is closed.

Tip: If the Create Copy option was used (when applying Effects to a Selected Object in the VIP window), you can use the Take Manager to switch between any appended (processed) copies, and the original (dry) audio file!

Extended Options for destructive effects

Here you can set some options for the application of destructive effects.

Destructive effect calculation in VIP Objects

In the upper part of the dialog you can choose between three ways of storing the calculated effect audio from applying destructive effects on VIP Objects. This data will be stored for "Undo" operations, when the checkbox "Create copy" in the dialog of the chosen effect is set.

Hint: Disable this checkbox only if you are really sure that you won't need to undo the calculation and you want to spare the time and disk space for creating the additional wave data.

1. "Append effect at original file." This was the usual way in earlier program versions. You obtain fewer different files and can quickly access the effect calculations according to a specific file. There are some disadvantages: when you are working with a 16bit-depth file, the effect is also stored in this resolution, which is undesirable when you want to benefit from the internal 32bit calculation depth. Furthermore, the file length is changed, which will cause problems when you use the Wave Project as a loop or in different VIPs.
2. "Write effect to an FX file (..._fx.wav)" The results of all effect calculations are stored in one separate file, named like the original Wave Project with the suffix "_fx". So the original Wave Project remains untouched. Additionally you've got the possibility to store the effect calculation in 32bit float format to remain the highest quality of the effect.
3. Finally, you can "Generate a new fx file for each effect". Then all destructive effects are stored in separate files with a running-number-suffix. Alternatively these files can also have detailed names.

Destructive effect calculation for Wave Projects

When calling a destructive effect from within a Wave Project, the above options are obsolete and the effect results are always inserted in the Wave Project. Nevertheless temporary files for undo actions are created, when "Undo" for Wave Projects and "Create Copy" are enabled.

When applying an effect on a Range in the Wave Project, it is possible to add a crossfade at the beginning and the end of the processed audio data between the original and effect audio. To do this, activate the according checkbox in the effect dialog "Create Crossfade". The length of the crossfades are also adjusted here.

Amplitude/Normalize → Normalize

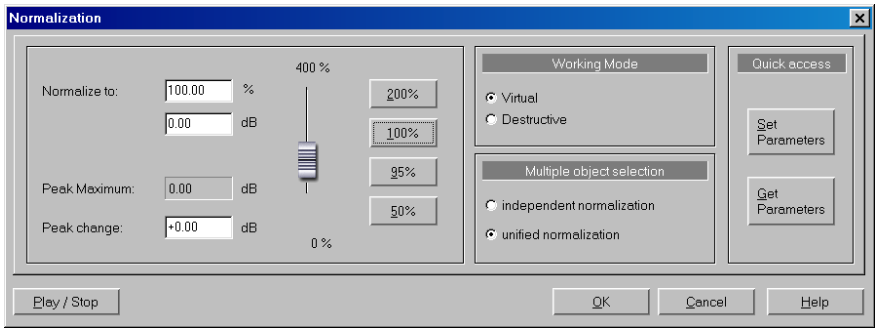
This function modifies the sample's overall amplitude.

The data is altered so that the maximum amplitude occurring in a specified Range is set to 100% (or any other value between 1-400%). Samplitude will first attempt to detect the maximum value, and relate it to the chosen percentage. All other values are then weighted with the new factor.

The Normalize function is designed to fully-modulate, or over-modulate, samples. It is often performed before converting a higher resolution sample to

a lower resolution. As the dynamic range of the low resolution file is reduced, it can still be fully-utilized by applying the Normalize function.
If you are working with sounds from a single instrument, you should set the factor to 100%.
If, however, your audio material features background percussion, for example, you will be able to over-modulate the sample to a range between 120% and 200%. This will only cut off the new percussion peaks. The same method allows you to alter the sound of natural instruments by over-modulating them.
As preparation for further physical processing, such as filtering, reverb, dynamic compression, etc., a level reduction of 50-70% is suggested. This should avoid clipping during post processing.

An important point to note: If the volume level is relatively low during recording, and the material is later normalized, the result will not be of the same quality as a full-range recording. As an example, if the volume level was only set to 50% of the possible range, the audio material will be in 15bit quality. Even normalizing the material to 100% will not change this aspect.



Normalize to: Allows you to set the target level for normalization. This can be done by typing the desired value into the data fields, adjusting the fader, or pressing the preset buttons for 50, 95, 100 or 200%. The value is displayed as a percentage and in dB. 0dB=100% (full Range). Normalizing to a value over 0dB results in digital distortion.

Peak maximum: displays the highest detected peak in the selected Range/Object.

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

Working Mode

Virtual: This function will perform real-time normalizing on the selected Objects. This is different than the destructive normalizing process, which restructures the audio file.
The real-time normalizing function looks for peaks in the Object, and adjusts the Object volume, so that the highest peaks precisely match the selected normalize level. You can return the Object to its original volume setting by

selecting "Reset", located below the volume fader in the Object Editor.
This option is not available when working with Wave Projects (Wave Editing)

Destructive: The audio material in the Wave Project is physically altered. Undo is only available when Undo is enabled, and the "create copy" flag is set.

Multiple selection

When multiple Objects are selected, there are two different techniques to normalize the Objects.

Independent normalization: Each Object is normalized according to its own maximum (peak) level.

Unified normalization: The maximum level is detected from all Objects, and each Object is normalized according to that value. This is the default behavior.

Quick Access

Set parameters: These buttons allow you to set a normalize value (e.g. 99%), and the normalize method for multiple selected Objects, when you access the Normalize function via the Quick Access dialog.

Get parameters: To reload these parameters in the dialog.

Shortcuts:
Key: *Shift + "N"*

Amplitude/Normalize → Normalize (quick access)

Quickly normalize a selected Range (Wave Projects), or selected Objects, (VIP) to a fixed value. This value can be defined in the Normalize dialog.
Normalization modes are always applied in accordance with the project type - i.e. Wave Projects in destructive editing mode are normalized "destructively" and Wave Projects in virtual Wave Editing mode (and VIP Objects) are normalized "virtually".

Key: *"N"*

Amplitude/Normalize → Fade In/Out

This option opens the Fade In/Out window. You can destructively apply fades (both in and out) to Wave Projects from this window.

Important note: Using this dialog, fades (both in and out) can be destructively applied to Wave Projects only. If you wish to destructively apply a fade to an Object in the VIP window, right-click on the Object and select Wave Editing. This will open the Wave Project that the Object refers to.

If you prefer to fade non-destructively, you can use the Volume Curves or Object Handle options in the VIP.

Fade Start in %: This parameter sets the level at the beginning of the fade as a percentage of the original level. To create a simple fade in, set this parameter to a value of 0. To create a simple fade out, set this parameter to a value of 100.

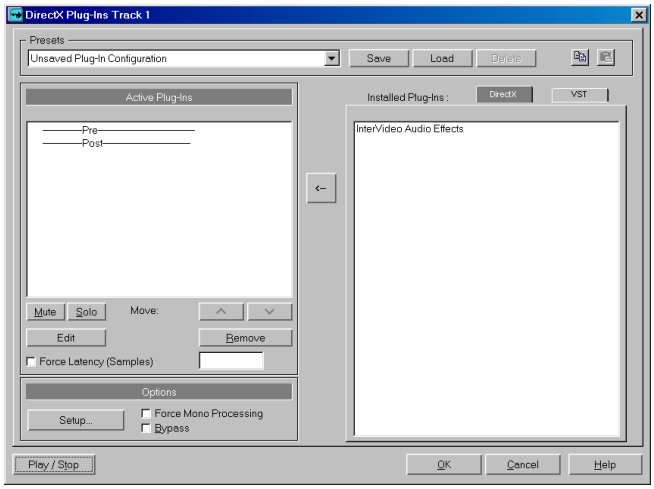
Note: You can adjust this parameter by clicking on one of the three "preset" percentage buttons (0, 50, 100), moving the slider, or by typing the value in the box.
Tip: Fade ins don't have to start at a level of 0, and end at a level of 100. Fade outs don't have to start at a level of 100, and end at a level of 0. The levels can be set anywhere between 0 and 200. Use the graphic display to help "visualize" the curve you are applying.

Fade End in %: This parameter sets the level at the end of the fade as a percentage of the original level. To create a simple fade in, set this parameter to a value of 100. To create a simple fade out, set this parameter to a value of 0.

Note: You can adjust this parameter by clicking on one of the three "preset" percentage buttons (0, 50, 100), moving the slider, or by typing the value in the box.
Tip: Fade ins don't have to start at a level of 0, and end at a level of 100. Fade outs don't have to start at a level of 100, and end at a level of 0. The levels can be set anywhere between 0 and 200. Use the graphic display to help "visualize" the curve you are applying.

Fade Curve: This parameter allows you to choose the type of curve used for the fade.

EXP.: Click on this button to select an exponential curve. The taper of the curve can be set with the slider.



LINEAR*:* Click on this button to select a linear curve. The taper of the curve can be set with the slider.

LOG.*:* Click on this button to select a logarithmic curve. The taper of the curve can be set with the slider.

Tip: By selecting the opposite number for a curve's taper, you can invert the curve.

Fade In: Click on this button to quickly set up a "simple" fade in. The fade's level will start at 0 and end at 100.

Fade Out: Click on this button to quickly set up a "simple" fade out. The fade's level will start at 100 and end at 0.

Important note: Whenever possible, it is recommended that you perform fades non-destructively within the VIP window by using the volume curves or Object handles. If you prefer to fade destructively, note that the Fade dialog does not have a Create Copy option. Don't panic, the Fade function is one of the Effects menu items which automatically creates an Undo if the Options > Program Preferences > Undo Definitions dialog has Undo enabled, for HD Wave Projects.

Shortcut: "F"

Amplitude/Normalize → Set Zero

Sample data values in a selected Range are set to zero (no data), eliminating noise and imperfections in a sample.

Example

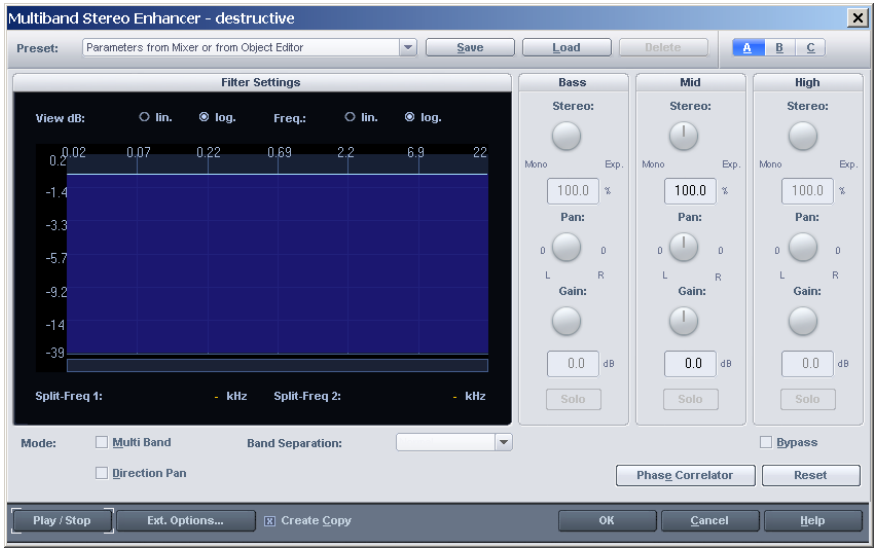
In between sentences in a spoken recording, there are inevitably background noises (e.g. pages ruffling). By marking this area and then overwriting it with silence (to zero), you can clean up your recording.

Switch Channels

This option allows you to switch the left and right channels of a stereo Wave Project.

Stereo Enhancer

This option opens the Multi-Band Stereo Enhancer window. In this window, you can expand or collapse the stereo image (in up to three separate frequency bands) of Wave Projects, or selected Objects in the VIP window.



Note: This option can only be applied to stereo audio files.

The Multi-Band Stereo Enhancer allows you to perform detailed modifications and corrections to the stereo image in three independent frequency bands. The signal is separated into the three frequency bands and the separate bands are processed individually.
The reassembly of the independent bands is 100% phase-neutral, thanks to the FIR Complement Filter technique – no discoloration of the frequency response is introduced. This method is superior to that of many other stereo enhancers, which work by destroying phase coherency.
Using multiple bands to affect the stereo image has many important advantages over image enhancers that destroy phase coherency - increasing the width of the mid-range, for example, prevents the ”typical” problems of muddy highs and lows when using such enhancers. These issues often show up due to phase cancellations. In addition, the decline of mono compatibility caused by an increase of the base width is limited to the treatment of a specific band.

Some Important Multi-Band Stereo Enhancer Applications

1. Reduction or expansion of the stereo base width.
2. ”Punchy” bass - through reduction of the base width in the bass range.
3. Control and correction of problems in the stereo image of a completed mix.

4. Relocation of the mono portion of a stereo recording within the panorama field (directional mixing). This can also be used to move a centered vocal track to the left or right in a stereo image.
5. 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.

In the upper left hand section of the dialog is the Filter section. The graphic display shows the approximate frequency response of the individual bands. The left axis shows the damping in –dB, the upper labeling the frequency in kHz. You can switch the according scales into linear or logarithmic mode. With the two sliders in Multiband mode you can adjust the split frequencies between the lower and middle (1) and the middle and high (2) band.
In the upper middle are the parameters for the stereo manipulation. To the right are the standard control options, radio buttons, and checkboxes for the different modes and the A/B/C switch for quick comparison of three dialog configurations.

Multi Band Mode

The algorithm works with three frequency bands if this option is checked.
The Multi Band mode is not automatically active when using the Enhancer in the Mixer Master section. This reduces the initial load on the CPU when selected in the Master section.

Band Separation

This setting determines the characteristics of the filter segment, and is used to increase separation.
Higher settings will produce the following:
1. The attack level of the filter curves is increased – the transition between two bands is smaller.
2. The absorption in the cross section increases: Setting Low: approx. 25-35 dB, Normal: approx. 35-45 dB, High: approx. 55-75 dB.
3. The coarseness of the frequency response of the individual bands is reduced. This is usually not a problem, as the coarseness of the individual filter bands is cancelled out, due to the complementary filter technique. The output signal is always reproduced without any coarseness.
4. The CPU load increases.

Direction Pan Mode

This switches between two panorama control modes.
If the option is checked, only the mono portion (the center signal) is considered when changing the panorama setting. The panorama control functions as a directional mix controller. This allows you to move previously centered vocals to the left or right channel. Any portion of the remaining audio signal that is not centered is unaltered.
Otherwise, all panorama controls operate conventionally – the complete stereo signal is changed (mono and the remaining left and right portions).

Stereo (Base Width)

The width value is set between 0 and 200. A setting of 0 means "Mono", 100 = "No Change" or "Stereo", and 200 = "Maximum Width" (Difference Signal). Depending on the correlation between left and right, a level boost can occur if the value of the fader setting is reduced. In extreme cases, a level boost of +3 dB may be introduced. This is the case if the left and right channels are identical (this corresponds to maximum correlation), and when setting the base width to 0 (Mono).
Increasing the base width to values above 100 decreases mono compatibility. If the base width is reduced, mono compatibility is maintained.

Gain

You can adjust the gain of each single band by +/- 6dB here.

Pan

The panorama for each of the bands is controlled with this knob (or horizontal slider). Damping values are shown in dB.
If the Direction Pan mode option is active, the Pan control functions as a directional controller. Only the mono portion (center signal) is considered.
This allows you to isolate centered vocals, and change their position within the stereo field. Any material outside the center remains unchanged.

Solo Mode

Selecting the Solo mode buttons allows previewing of individual frequency bands. This greatly simplifies the process of changing filter parameter settings. As an example, you can isolate a specific frequency range of a mix, and change the aspect of the stereo image for the range.

A/B/C

To compare different dialog configurations, select a different radio button. Can also be used as a bypass function, if one of the configurations is left at the default setting.

Phase Correlator

Opens the Phase Correlation Display. This is especially useful in Solo mode. The base width and panorama setting for each frequency band can be visually checked.

Playback

Click on this button to hear a real-time audition of the processed audio.

Save, Load, Delete Setup

This allows you to save, load, and delete configuration settings for the dialog. The default file extension is *.ste.

OK
The algorithm is applied to the selected Range or VIP Object.

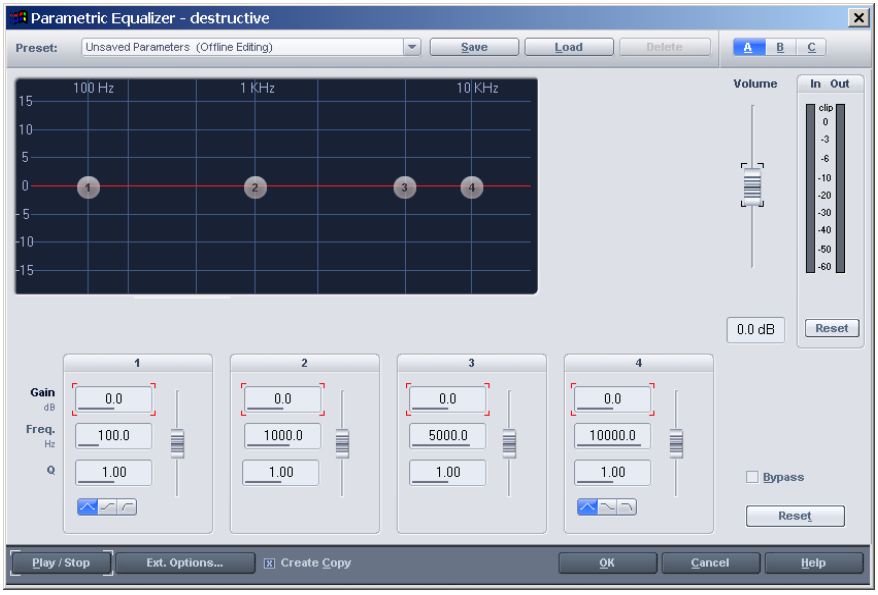
Cancel
The dialog is closed without applying any changes.

Invert Phase

The sample data within the selected Range is inverted along the amplitude axis. This phase inversion means that negative values become positive, and vice versa. This function is reversed by simply reusing the command.
The "Invert" function allows samples with different phases to be matched. Along with the available mixing functions (which are, from a mathematical viewpoint, adding functions), you can actually subtract samples by applying this function to the selected sample.

Parametric Equalizer....

This dialog contains a 4-band parametric equalizer. You can activate filters in four, freely-selectable frequency ranges, in order to adjust the sound of a sample. You can make broad (wide-band) frequency adjustments for both high and low pass ranges, plus fine (narrow-band) corrections of specific frequency ranges.
Note that the Mixer window (Shortcut: "M") also offers one equalizer processor per channel. This processor will operate in real time and the audio material will not be altered. (Non-destructive editing.)



1..4 The EQ bands are arranged in groups of buttons, controlling frequency, bandwidth (Q) and gain (volume). Each of these parameters can be adjusted by

moving the corresponding knob, or thru numerical input in the data fields. The EQ can also be adjusted in the frequency response curve above. Drag the according ball for an EQ band to form the curve.

Type: For the first and fourth EQ band (typically used for the lowest and highest frequency bands), you can select a lowpass, shelving or bandpass filter curve.

Freq.: The frequency of the individual filters can be adjusted between 10 Hz and 24 kHz. As each frequency is selectable, several filters can also be set to the same values, to achieve a more pronounced effect.

Q: The bandwidth of each individual filter can be adjusted between 10 Hz and 10 kHz. Once again, overlapping filter frequencies are possible.

Gain Db: These knobs determine the amount that the filter level (volume) is increased or decreased (+/- 20 dB). A setting of 0 deactivates the filter, and does not consume processing power.

Volume: You can adjust the output gain (volume) with this fader. Your filtering results in a low volume (i.e. makeup gain).

In/Out Meters: These meters indicate the levels of the audio input and output. Reset: Resets the meters, in cases where clipping has occurred. (Turns off the red clip indicator)

A, B, C: You can choose between three different settings with these checkboxes. This is especially helpful if you want to experiment with settings, and compare them. To switch between setups, simply select another radio button and adjust settings to your taste, then select another radio button a.s.o. The settings will be stored with the project and can be saved as a group with the Save button.

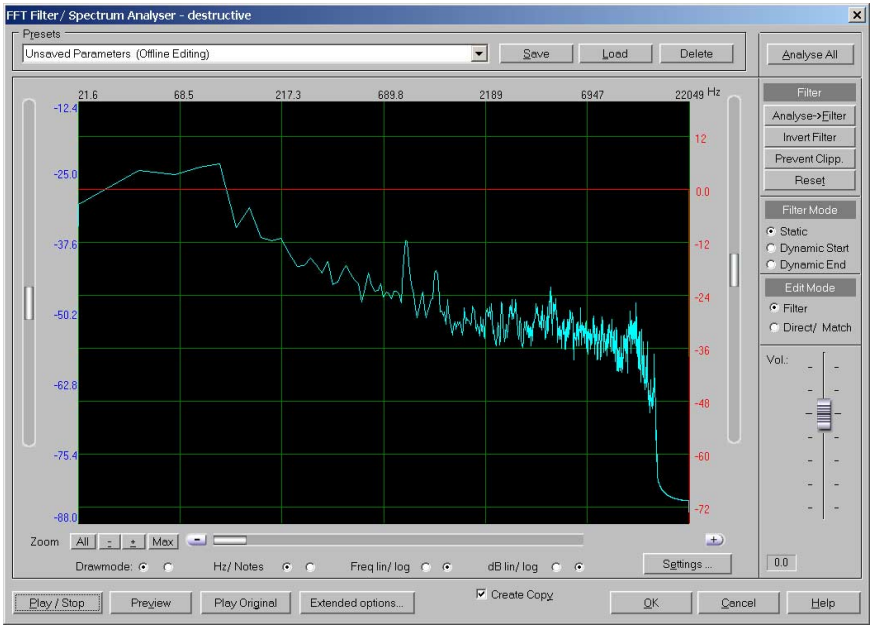
Bypass: This option bypasses the EQ entirely, allowing an easy comparison between the processed and unprocessed signal.

Create Copy: As with most of the other Effects menu items, this important option determines whether an Undo will be created - prior to processing the file. As always, we strongly recommend that you leave this option enabled.

Play/Stop: This button activates the real-time preview, allowing you to audition the EQ settings, prior to applying the process.

FFT Filter/Analyzer...

Choose this option to open the FFT Analyzer/Filter window. This window allows you to destructively apply FFT Filtering to Wave Projects or to selected Objects in the VIP window. The FFT Filter is one of Samplitude's most powerful tools, offering unique draw options that allow you to easily redraw the frequency response of the selected Range. This can be invaluable when dealing with equalization issues, for repairing audio, or for precision mastering.



So...what is it?

The FFT Filter/Analyzer combines a FFT (Fast Fourier Transform) spectrum analyzer with a FFT Filter. In other words, a Spectrum Analyzer to view the frequency response of the audio, and a graphic EQ that allows you to draw the desired changes into the frequency response graphic.

What can I do with it?

There are literally a million and one possible uses for this processor. In fact, a manual as thick as the one you have in your hand could easily be written on this subject alone! With that in mind, we'll cover some of the more common, practical uses.

This manual assumes that you, the reader, have reasonable knowledge of the principles and terminology of audio spectrum analysis.

1. View the frequency response of a mix, and make desired adjustments
2. Sub-bass enhancement or reduction (of rumble or plosives)
3. Adjust the amplitude of an individual instrument or note
4. Capture the frequency response from a recording, and apply it to another recording (convolution)

5. Crossfade between two separate filter curves

Tip: If you wish to delve further into the theory behind the FFT-Filter/Analyzer, we suggest you find an advanced text, specific to FFT.

Curve Display

Three different curves (each a separate color) can be displayed by the graph. A total of nine curves (three groups of three) are available for use.
The first group of three curves (yellow, red, blue) is used in the Static mode, the second group is used for the start point of the dynamic filtering (Dynamic Start), and the third is used for the end point of the dynamic filtering (Dynamic End).

Blue: The blue curve represents the original frequency response - UNTIL changes have been made. Once changes to the frequency response curve have been made, the blue curve represents the resulting frequency response. When using the Direct/Match Edit mode, you can re-draw this curve, in order to make absolute changes to the frequency response. In other words, the curve you draw IS the new frequency response curve.

Yellow: When making changes to the frequency response curve, the yellow curve will appear, indicating the original frequency response. The yellow curve only appears when using the Direct/Match Edit mode, or when using the Filter Edit mode - with curve numbers set to 3 (under the *Settings* > *Draw Settings* > *Curve Number* option).

Red: The red curve is centered (flat) at 0dB, and indicates relative changes (i.e. the gain changes made, to achieve the desired response) to the frequency response curve. You can re-draw this curve in order to make relative changes to the frequency response.

Faders

Left (large) fader: Use this fader to adjust the amplitude scale of the blue and yellow curves. The amplitude is shown (in dB units) to the right of the fader.

Right (upper) fader: Use this fader to adjust the amplitude scale of the red curve. The amplitude is shown (in dB units) to the left of the fader.

Right (lower) fader: Use this fader to adjust the overall level of the FFT Filter, without altering the frequency response curve. The amplitude change is shown (in dB units) beneath the fader.

Zoom

All: Click on this button to quickly zoom out all the way. If you click on the button a second time, the graphic display will return to the previous zoom level.

Out: Click on this button to zoom out by 1 level (there are 10 zoom levels).

In: Click on this button to zoom in by 1 level (there are 10 zoom levels).

Max: Click on this button to quickly zoom in all the way. If you click on the button a second time, the graphic display will return to the previous zoom level.
There are a total of 10 zoom levels available.

Note: Once zoomed-in, you can use the scroll bar (beneath the graphic display) to scroll the display. The current frequency range is shown (in Hz) above the graphic display.

Draw Mode (below the zoom buttons)
In the first mode (left radio button), the frequency response is displayed as individual "partials" (useful for editing the low end of the frequency response). In the second mode (right radio button), the frequency response is displayed as a curve.

Hz/Notes (to the right of Draw mode)
This option allows you to display the graphic in Hz (frequency) or musical notes. Frequency can only be displayed as musical notes when the Freq lin/log parameter is set to log.

Freq lin/log (to the right of Hz/Notes)
Linear display of frequency provides a more detailed view of the higher end. Logarithmic display of frequency provides a more detailed view of the lower end.

dB lin/log (to the right of Freq lin/log)
Linear display of amplitude provides a more detailed view of the higher end. Logarithmic display of amplitude provides a more detailed view of the lower end.

Filter Mode
Static: Select this mode to affect the entire Wave Project or selected Object. You should always use this mode, unless you intend on crossfading between two frequency response curves.

Dynamic start: When a crossfade between two frequency response curves is desired, use this mode to adjust the first (start) curve. The FFT Filter will start with this curve, and perform a crossfade until it ends on the second (end) curve.

Dynamic end: When a crossfade between two frequency response curves is desired, use this mode to adjust the second (end) curve. The FFT Filter will

start with the first (start) curve, and perform a crossfade until it ends on this curve.

Edit Mode

Filter: Select this mode to re-draw the red curve – i.e. use this mode to make relative adjustments to the frequency response curve.

Direct/Match: Select this mode to re-draw the blue curve - i.e. use this mode to make absolute adjustments to the frequency response curve. The curve you draw IS the new frequency response curve.

Analyze all

Click on this button to have the FFT Spectrum Analyzer calculate the frequency response of the Wave Project or selected Object. Note that this option only applies when the Filter mode is set to ”Static”.

Reset

Click on this button to reset all curves to their default (0dB) position.

Analyze > Filter

Click on this button to transfer the frequency response curve (of the current Wave Project, or selected Object) to the red curve. This allows you to quickly apply the frequency response of one recording to another.

Invert Filter

Click on this button to invert the frequency response curve.

Prevent Clip.

If the audio distorts or clips after adjustment of the frequency response curve, click on this button to instantly readjust the curve (in a way that prevents clipping).

Create Copy

As with most of the other Effects menu items, this important option determines whether an undo will be created prior to processing the file. As always, we strongly recommend that you leave this option enabled!

Play/Stop

Click on this button to hear a real-time stereo preview of the filtered audio.

Preview

Click on this button to hear a calculated (processed) preview of the filtered audio. This option should be used if your machine isn’t fast enough to allow real-time playback.

Tip: You can determine the length of the preview by pressing the "Y" key, and entering the desired number of seconds in the Program Options > Preview Time parameter box (middle of the window).

Play original
When using the *Test/Preview* audition functions, click on this button to play the original (dry) audio. This allows you to quickly compare processed vs. unprocessed audio.

Settings
Click on this button to open the Advanced Settings window.
This dialog allows you to adjust the factors used in calculating the analysis of the audio. In most cases, the default values are fine for most work, but you may wish to experiment with the settings, should you have projects or work which require(s) greater accuracy.
To improve accuracy in the lower frequency ranges, a higher resolution is needed (starting at 8192). The accuracy of the analysis (in Hz, at a sample rate of 44.1 kHz) is listed in the table below:

Resolution	Accuracy in Hz
256	172
512	86
1024	43
2048	21.5
4096	10.7
8192	5.3
16384	2.7
32768	1.35

High resolutions (8192) are only needed for low frequency ranges, or specific (fading) harmonics. Generally, values of 1024 or 2048 are ideal. Values below 1024 rarely yield useful results.
Two important suggestions:
1. 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.
2. When previewing the filter settings, and working with high resolutions in real time, please ensure that the *Test* Buffer setting (*File > Options > System*) is the same as, or slightly higher than, the Resolution setting (in the Advanced Setting dialog of the FFT Analyzer). Real-time previews are not possible unless this is correctly set.

If the *Test* audio is breaking up/popping/crackling, this is probably an indication that you need to adjust the buffer or resolution settings.

Settings > Analysis Parameters

Analysis Precision High: When analyzed, the audio is separated into individual blocks. This mode overlaps the blocks by a 50% ratio, providing a more accurate representation of short sounds (percussion, etc.). Calculation time is doubled when this setting is selected.

Analysis Precision Normal: When analyzed, the audio is separated into individual blocks. This mode appends the blocks, without overlapping them.

Analysis Accuracy Noise: This is the preferred option for analyzing audio that has a less distinct pitch (e.g. percussion or cymbals)

Analysis Accuracy Tones: This is the preferred option for analyzing audio that has a more distinct pitch (e.g. guitar or flute)
When working with completed songs (i.e. a file which has already been mixed), the choice of setting used here will depend on the type of material. You will need to use your own judgment in such cases.

Channels Left: When working with stereo files, select this option to limit analysis to the left channel.

Channels Right: When working with stereo files, select this option to limit analysis to the right channel.

Channels Both: When working with stereo files, select this option to analyze both channels.

Analysis Time(s): This parameter allows you to define the number of seconds that the audio will be analyzed by the FFT Analyzer/Filter.
This setting only affects the display of frequency response in "Static" mode.

Note: Every time an analysis parameter, or the resolution is changed, the frequency response is re-calculated. For instance, if you have previously used the "Analyze All" button to calculate the response for the complete Range, you will need to reuse the function. This is because the re-calculation (due to the aforementioned changes) will overwrite the previous analysis results. In this situation, the "Analyze All" button is re-enabled.

Settings > Filter Parameters

Precision: This parameter allows you to specify the quality level that the FFT Filter will use when processing the audio file. Low, Normal, and High are the choices; High offering the best quality, but also taking the longest time for processing.

Precision Real Time: This parameter allows you to specify the quality level used for real-time previewing of the FFT Filter. Test, Low, Normal, and High

are the four choices; with High offering the best quality, but also requiring the most computing resources.

Tip: If your machine isn't fast enough to allow a real-time preview at the High setting, try one of the lower settings. The quality of the preview will be slightly lower, but that won't affect the quality level of FFT Filtering on the audio file.

Max Range (dB): This parameter sets the maximum amplification range for the red curve.

Min Range (dB): This parameter sets the minimum amplification range for the red curve. If frequencies are to be completely faded out, set the Min Range parameter to a low value (e.g. –120dB).
Note: The fader to the right of the graphic display can be used for fine tuning.

Threshold: When re-drawing the blue curve (Direct/Match Edit mode), only frequencies that were originally louder than this threshold setting will be changed. This makes adjusting the levels of partials easier (when using the "Partial" Draw mode).

Note: The fader to the right of the graphic display can be used for fine tuning.

Settings > Draw Settings
These settings affect the way the frequency response curves are displayed.

Draw Grid: This parameter allows you to toggle the Grid On and Off for the graphic display.

Curve Numbers: This parameter allows you to determine the number of curves shown in the graphic display. The choices are 1, 2, or 3.

Settings > Dynamic Filter Mode
Three different modes are available when using dynamic filtering.

Direct: A simple crossfade between the two curves is performed.

Circle/Direct: A crossfade is performed between the two curves, and the start curve is moved towards the higher or lower frequencies. If you selected a band pass filter as the start curve, the middle frequencies are changed in time (filter sweep). The highest value of the start and end curves are determined, and a range for the "sweep" is set.

Note: If in the Filter Edit mode, you will first need to edit the red curve. If the complete filter curve is at 0 dB, no highest value is available for calculation.

Circle: The start curve is moved in this mode. The highest and lowest values of the start and end curves determine the "sweep" range. The second curve is

only used to determine the end of the sweep, and otherwise, has no influence on the end result.

Note: If in the Filter Edit mode, you will first need to edit the red curve. If the complete filter curve is at 0 dB, no highest value is available for calculation.
Tip: Use a broadband (noise) signal to experiment with the different settings for the Dynamic Filter mode. This will allow you to easily hear how the dynamic filter works.

Settings > Resolution

This parameter determines the number of individual filter bands used when calculating the filter curves. The actual number of filter bands used is one half of the value specified here.

Note: Resolutions above 8192 are only needed when performing detailed work with low frequency Ranges, or for fading individual "partials". For all other uses, a value of 2048 or 4096 is usually sufficient.

Settings > Save Setup

Click on this button to save the current parameter settings and filter curves.

Settings > Load Setup

Click on this button to load a previously-saved set of parameters and filter curves.

Note: Once you have selected the settings, you can close the Advanced Settings window. The settings will be kept for the remainder of your current Samplitude session, even if you haven't saved the settings via the "Save Setup" button. When you restart Samplitude, the default settings will be loaded when the FFT Filter is opened if the settings were not saved as a preset.

To view the frequency response of a mix and make desired adjustments:

1. If the FFT Analyzer/Filter is to be applied directly to a Wave Project, ensure that it is the currently active window. If the FFT Analyzer/Filter is to be applied to an Object in the VIP window, ensure that the Object is selected. (Click on the Object, in the lower half of the track, to select it.)
2. If the FFT Analyzer/Filter is to be applied directly to a Wave Project, mark a Range in the Wave Project window. If the FFT Analyzer/Filter is to be applied to a selected Object in the VIP window, proceed to step 3.
3. Select the *Effects > FFT Filter/Analyzer menu* option, and the FFT Analyzer/Filter window will open.
4. In the lower-left corner of the window (for Draw mode), select the second "Curve" mode. The frequency response will be displayed as a curve.
5. Click on (i.e. enable) the *Direct/Match* option (on the right side of the window - Edit mode).

6. Click on the "*Analyze All*" button. The entire marked Range, or selected Object, will now be analyzed for frequency content. When the analysis has finished, the frequency content will be displayed as a blue curve.
7. You can now use the mouse to re-draw any section of the blue curve - i.e. if the graph shows a tall "spike" at 1000Hz (and the audio sounds a bit harsh), use the mouse to re-draw the spike (make it shorter). The FFT Analyzer/ Filter is perfect for taming strident frequencies.
8. Click on one of the *Test* buttons to hear a real-time preview of your frequency curve changes.
9. Repeat steps 6 and 7, until you are satisfied with the results. If you'd like to start the re-draw process over again, click on the "*Reset*" button, and the curve will return to its original analyzed state.
10. Click on the *OK* button to "write" the FFT Filter process to the Wave Project or selected Object, and to close the FFT Analyzer/Filter window.

Enhancing or reducing sub-bass:

1. If the FFT Analyzer/Filter is to be applied directly to a Wave Project, ensure that is the currently active window. If the FFT Analyzer/Filter is to be applied to an Object in the VIP window, ensure that the Object is selected. (Click on the Object, in the lower half of the track, to select it.)
2. If the FFT Analyzer/Filter is to be applied directly to a Wave Project, mark a Range in the Wave Project window. If the FFT Analyzer/Filter is to be applied to a selected Object in the VIP window, proceed to step 3.
3. Select the *Effects > FFT Filter/Analyzer menu* option, and the FFT Analyzer/ Filter window will open.
4. In the lower-left corner of the window (for Draw mode), select the first "Partial" option.
5. Click on (i.e. enable) the *Direct/Match* option (on the right side of the window - Edit mode).
6. Click on the *Settings* button to open the Advanced Settings window. In the upper-right corner of the window, set the resolution to 16384. Close the Advanced Settings window.
7. Click on the "*Analyze All*" button. The entire marked Range, or selected Object, will now be analyzed for frequency content. When the analysis has finished, the frequency Content will be displayed as a blue curve.
8. You can now use the mouse to increase or decrease the level of each partial between 10Hz and 50Hz. *Warning: You can potentially damage your monitors when excessively boosting sub-bass frequencies. Keep your monitor levels low when experimenting!* Click the mouse just to the right of each "partial', in order to adjust its level.
9. Click on one of the *Test* buttons to hear a real-time preview of your frequency curve changes.
10. Repeat steps 8 and 9 until you are satisfied with the results. If you'd like to start the re-draw process over again, click on the "Reset" button, and the curve will return to its original analyzed state.
11. Click on the *OK* button to "write" the FFT Filter process to the Wave Project or selected Object, and to close the FFT Analyzer/Filter window.

Adjusting the amplitude of an individual instrument or note:

1. If the FFT Analyzer/Filter is to be applied directly to a Wave Project, ensure that it is the currently active window. If the FFT Analyzer/Filter is to be applied to an Object in the VIP window, ensure that the Object is selected. (Click on the Object, in the lower half of the track, to select it.)
2. If the FFT Analyzer/Filter is to be applied directly to a Wave Project, mark a Range in the Wave Project window. If the FFT Analyzer/Filter is to be applied to a selected Object in the VIP window, proceed to step 3.
3. Select the *Effects > FFT Filter/Analyzer menu* option, and the FFT Analyzer/Filter window will open.
4. In the lower-left corner of the window (for Draw mode), select the first "Partial" option.
5. Click on (i.e. enable) the *Direct/Match* option (on the right side of the window - Edit mode).
6. Click on the "*Analyze All"* button. The entire marked Range, or selected Object, will now be analyzed for frequency content. When the analysis has finished, the frequency content will be displayed as a blue curve.
7. Click on the "*Max* Zoom" button to zoom the graphic display all the way in.
8. If you wish to adjust the amplitude of an instrument, select the first 'Hz' option for the Hz/Notes parameter. Frequency will be displayed in Hertz (Hz). If you wish to adjust the amplitude of a note, select the second "Note" option for the Hz/Notes parameter. Frequency will be displayed as musical pitches (notes).
9. To adjust the amplitude of an instrument, find its fundamental (strongest) frequency range, and re-draw the "partials" in that area (higher – to increase amplitude, or lower – to decrease amplitude). To adjust the amplitude of a note, find the partials in the note's frequency range and re-draw them (higher – to increase amplitude, or lower – to decrease amplitude).
10. Click on one of the *Test* buttons, to hear a real-time preview of your frequency curve changes.
11. Repeat steps 9 and 10, until you are satisfied with the results. If you'd like to start the re-draw process over again, click on the *Reset* button, and the curve will return to its original analyzed state.
12. Click on the *OK* button to "write" the FFT Filter process to the Wave Project or selected Object, and to close the FFT Analyzer/Filter window.

Capturing the frequency response from a recording and applying it to another recording:

1. If the FFT Analyzer/Filter is to be applied directly to a Wave Project, ensure that it is the currently active window. If the FFT Analyzer/Filter is to be applied to an Object in the VIP window, ensure that the Object is selected. (Click on the Object, in the lower half of the track, to select it.)
2. If the FFT Analyzer/Filter is to be applied directly to a Wave Project, mark a Range in the Wave Project window. If the FFT Analyzer/Filter is to be applied to a selected Object in the VIP window, proceed to step 3.

3. Select the *Effects > FFT Filter/Analyzer menu* option, and the FFT Analyzer/ Filter window will open.
4. Click on (i.e. enable) the *Direct/Match* option (on the right side of the window - Edit mode).
5. Click on the "*Analyze All*" button. The entire marked Range, or selected Object, will now be analyzed for frequency content. When the analysis has finished, the frequency content will be displayed as a blue curve.
6. Click on the *Analyzer > Filter* button, and the analyzed frequency response curve will be transferred to the red curve. Click on (i.e. enable) the *Filter* option (Edit mode). The red curve (representing the analyzed audio's frequency response) is now visible.
7. Click on *Cancel* to close the FFT Analyzer/Filter window.
8. If the red (analyzed) frequency response curve is to be applied directly to a Wave Project, ensure that it is the currently active window. If the red (analyzed) frequency response curve is to be applied to an Object in the VIP window, ensure that the Object is selected. (Click on the Object, in the lower half of the track, to select it.)
9. If the red (analyzed) frequency response curve is to be applied directly to an Wave Project, mark a Range in the appropriate Wave Project window. If the FFT Analyzer/Filter is to be applied to a selected Object in the VIP window, proceed to step 10.
10. Select the *Effects > FFT Filter/Analyzer menu* option, and the FFT Analyzer/ Filter window will open.
11. Click on one of the *Play* buttons, to hear a real-time preview.
12. Click on the *OK* button to "write" the FFT Filter process to the Wave Project or selected Object, and to close the FFT Analyzer/Filter window.

Crossfading between two filter curves:

1. If the FFT Analyzer/Filter is to be applied directly to a Wave Project, ensure that it is the currently active window. If the FFT Analyzer/Filter is to be applied to an Object in the VIP window, ensure that the Object is selected. (Click on the Object, in the lower half of the track, to select it.)
2. If the FFT Analyzer/Filter is to be applied directly to a Wave Project, mark a Range in the Wave Project window. (Click and drag in the appropriate Wave Project window to mark a Range.) If the FFT Analyzer/Filter is to be applied to a selected Object in the VIP window, proceed to step 3.
3. Select the *Effects > FFT Filter/Analyzer menu* option. The FFT Analyzer/Filter window will open.
4. In the lower left corner of the window (for Draw mode), select the second "Curve" mode. The frequency response will be displayed as a curve.
5. Click on (enable) the *Filter* option.
6. Click on (enable) the *Dynamic Start* option.
7. The currently displayed blue curve represents the frequency response at the beginning of the audio file. On the right side of the window under Filter mode, click on (i.e. enable) the *Dynamic End* option.
8. Re-draw the red curve (currently centered at 0dB) to make some relative changes to the frequency response.

9. Click on one of the *Test* buttons, to hear a real-time preview of the filter curve crossfade.
10. Repeat steps 8 and 9 until you are satisfied with the results. If you'd like to start the re-draw process over again, click on the *Reset* button, and the curve will return to its original analyzed state.
11. Click on the *OK* button to "write" the FFT Filter process to the Wave Project or selected Object, and to close the FFT Analyzer/Filter window.

Problems and Solutions regarding the FFT Filter Analyzer

1. Slight tremolo or rough sounding audio, delay effects:
This can happen when the "Test1" setting is selected in the Precision section of the Advanced Settings dialog. Use a different Precision setting.

2. Modulation Effects when using extreme filter settings for low frequency ranges:
This is also caused by the "Test1" setting, in the Precision section of the Advanced Settings dialog. Selecting a different setting may solve this problem.

3. Problems with the real-time preview when using higher resolutions:
Ensure that the File > Preferences > System > Real Time buffer setting is either the same as, or higher than that of, the Resolution setting found in the Advanced Setting dialog of the FFT Filter.

4. You are unable to completely eliminate specific ranges of a spectrum:
Ensure that you are using the Filter Edit mode, and reduce the dB Min setting to a value between –100 to –120 dB.

5. Differences between filtering and analysis:
If filter processing has occurred, and the frequency response of the edited Range is analyzed a second time, the new analytical process does not coincide with the corrected frequency response (the blue curve) before processing. Why not? The responses will closely resemble each other if the Analyze All, and the Analyze Precision "High" settings are used prior to the filtering process, and for the new analysis.

6. Unexpected resonance-like artifacts are introduced when filtering with broad band-passes:
Reduce the angle of the curves by omitting vertical lines when drawing the curves.

7. The morphing results for the dynamic filtering are uneven when using the Dynamic Start or End filter mode:
Use the highest Filter Precision setting.
The Direct dynamic filter mode reduces this problem when using smaller Resolution values. You may need to experiment with different settings for the other dynamic filter modes.

8. The results of the calculation (OK), or the non-real-time preview function (Preview) are somewhat different to the real-time previewing (Test):
If different settings are used in the Filter Precision and Filter Precision Real Time options (in the Settings dialog), this may result in slight differences in processing. Take note of these settings when working with the FFT.

Finally, when working with the FFT Filter, work carefully and patiently. Remember to keep track of all settings to avoid any confusion or inaccuracies, and remember to make sure that your Undo options are enabled!

Dynamics (Overview)

Overview concerning dynamic manipulation

Three modules are available for dynamic processing:

Dynamics: A simple, quick dynamic module, offering a compressor, expander, gate and limiter. This module uses an extremely efficient algorithm, making it very light on CPU resources.

Advanced Dynamics: The Advanced Dynamics module is a more extensive tool, which combines a classic dynamic module (compressor/expander/gate), and an approximating limiter. The output from this module is an unbiased, optimally-modulated signal with defined volume.
As opposed to the mode of operation used by the Dynamics module, the Advanced Dynamics module is controlled by the characteristic line, rather than setups. This approach allows free combinations of operating modes.
Parameter adjustments are possible through direct input, or graphic adjustment of the characteristic line.
Level detection is performed in (selectable) PEAK, RMS or "fast" modes. In "fast" mode, computer processing requirements are minimal, but the mode of operation differs from "classic" dynamic modules. The RMS mode mirrors the dynamic behavior of analog dynamic modules. A more lively sound is achieved in PEAK mode.

Multiband Dynamics: The most extensive dynamic processing options are afforded by the Multiband Dynamics module. The primary advantage of dynamic manipulation in several frequency bands (compared with a standard dynamics processor) is that "pumping", and other undesirable side effects, are drastically reduced. As an example, a level peak in the treble won't force the entire signal down in the bass Range, allowing you to focus on the processing of individual frequency Ranges.

Which module should be used, and when?

Which dynamic module should be used, and when, depends on how focused the change needs to be. A "Dynamics" module is incorporated into each mixer channel, and offers extremely light CPU use and simplicity of operation. Its inclusion on each channel makes the adjustment of overall track dynamics quick and efficient.

The sound of an Object, on the other hand, must often be modified very specifically. For example, the dynamics of an instrument can differ, due to the change in playing style between the verse and refrain. This is usually desirable, but can be smoothed or emphasized through precise use of compression. The use of the Advanced Dynamics module is advisable for precise changes to the dynamic behavior of the source material, as in our example.
The Multiband Dynamics module is generally used to balance the entire mix, in the master channel. One problem which may occur is clipping, resulting from the volume increase of the multiband compressor. To avoid this issue, it’s often necessary to insert a limiter which will then smooth all overly-hot levels to the limiter threshold. If using multiband dynamics during CD mastering, it’s usually better to use the processor in the Object Editor. This allows you to adjust the processor to suit each song (i.e. each Object, in this situation, would point to a file that was a full stereo mix of a song).
It should be noted that experienced mastering engineers are often cautious about using multiband dynamics. Actually, let’s shorten that sentence: experienced mastering engineers are often cautious, or to put it another way - *”there are no old bold mastering engineers”*. Are you starting to see where this is heading? Multiband dynamics processors have not always been readily available, although a variant known as side chain dynamics processing (e.g. de-essing) has been popular since the dawn of compression and EQ. Mastering engineers are conservative by nature, generally using as little compression as possible. (There are, of course, always exceptions!). Early multiband compressors were not phase coherent, resulting in coloration of the phase response. In some cases, the costs outweighed the benefits.
Over recent years, advances in DSP algorithms and computing power have resulted in a proliferation of powerful multiband dynamics processors. These are capable of rectifying mastering problems which were previously difficult to control. Samplitude’s phase-coherent multiband dynamics processor represents the pinnacle of this new technology, but is capable of destruction if used incorrectly or excessively. Some of the world’s most famous mastering engineers blame the current ”inappropriate hyper compression” trend (where every CD is mastered as loud as possible) on the advent of software/hardware ”Finalizers” and multiband compressors. While this claim may be at least partly true, the processors are not to blame - the person who controls the processor determines the product.

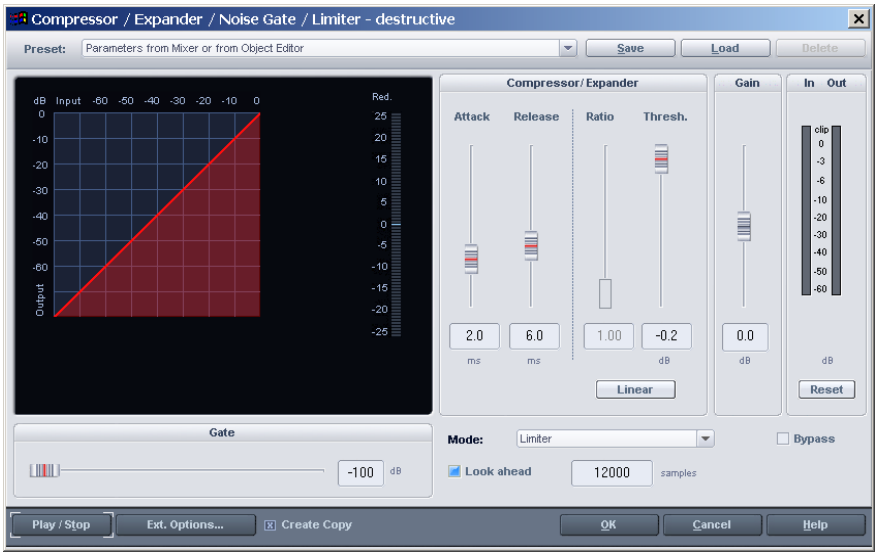
Note: Sometimes it is very difficult to avoid hyper compression, as the person who pays your engineering wages may be demanding the “loudest CD on the radio!”. Don’t worry, we understand. Just do your best to keep the dynamics as vibrant as possible.

Samplitude’s multiband dynamics processor is exceptionally powerful, and although very capable in any situation, it should only be used when actually needed. If multiband compression is *required* when working on a multi-track mixdown, you should first see if the audio can be isolated to one or two tracks. If so, try using the Multiband dynamics within the Object Editor for those

tracks/Objects, rather than inserting the processor over the entire mix. When mastering, carefully consider the reason for inserting the Multiband Dynamics module. Be patient when adjusting the parameters, and be sure that your chosen settings actually ”enhance” your mix.
In some cases, you may find that the advanced dynamics processor is more appropriate for the project. In a mastering situation, this would be termed ”bus compression”, as the processor is inserted over the stereo output buss. Put simply, the entire frequency spectrum is processed.
The advanced dynamics processor emulates the wonderful sound of vintage bus compressor/limiters, which shaped the sound of many famous recordings. On the other side of the coin, you can also use this processor to seriously mangle your audio into new, unexpected shapes. (Multimedia creators and sound designers take note!). It’s really a matter of knowing what your audio needs, then patiently adjusting the parameters until you strike gold.
When using multiband dynamics, it may be beneficial to insert the Advanced Dynamics module (set as a limiter) after the multiband. If you’re close to running out of DSP power (CPU resources), try using the standard Dynamics module (set as a limiter) instead.
When CD mastering, it’s recommended that multiband dynamics be used on a per-Object (CD track) basis. This allows precise control over the individual CD tracks, with each having different
multiband dynamic settings.
The following table indicates where the various dynamic modules are available:

Dynamics (Compressor/Expander....)

Choose this option to open the Compressor/Expander/Noise Gate/Limiter window. This window allows you to destructively apply dynamics processing to Wave Projects, or selected Objects in the VIP window.



The Compressor processes the dynamics of an audio sample. The dynamics algorithms ”look ahead” to eliminate peak distortions, or other artifacts. The graphics display always indicates the actual dynamics curve. The additional alternative uses as an expander, noise gate and limiter are variations on the compression algorithm.

Note: The Mixer (shortcut – “M”) and the Object Editor also contain a compressor for each channel or Object. These compressors operate in real time within the virtual environment – the audio material on the hard disk is not altered (i.e. non-destructive processing).

Mode

The track’s dynamics processor can function as a compressor, limiter, expander, gate, or distortion device. Select the appropriate mode in the Mode listbox to apply the specified type of dynamics processing.

Note: Proper use of dynamics processing is an art form in itself – it is beyond the scope of this manual to go into great detail about the use of dynamics processing in your Samplitude projects. Great care should be taken when using compressors, as the effect of removing dynamics from your audio can result in the loss of all excitement and life from perfectly good recordings! We have assumed that the you have a general familiarity with dynamics processing, and the terms/methods associated with such processes.

The dynamics algorithm is similar to that of a “classic” compressor.
The dynamic of a frequency band is limited, loud sections are reduced (i.e. compressed), and soft audio sections maintain their level. This setting is used if the compression is to result in a coloration of the audio material without causing the volume to rise.
The compression ratio is configured with the ratio control, the threshold with the threshold setting, and the envelope is controlled with the Attack and Release settings.

Comp Max: This mode is exactly the same as the compressor, but the output is normalized to 0dB.The dynamics of a frequency band are limited, loud passages stay loud, and soft audio sections become louder. This setting is often used to add ”presence” to audio material, making it more prevalent, or to increase loudness. Other terms for this type of processing are “Maximizer” or “Loudness Enhancer”.

Limiter: This mode limits the dynamics of audio to a predefined threshold. Loud sections (i.e. above the threshold) will be reduced, but soft sections (below the threshold) are unaffected.
The ratio is automatically set to infinity when in this mode.

Limiter 100: This mode is exactly the same as the limiter, but the output is normalized to 0dB.

The ratio is automatically set to infinity when in this mode.

Expander: The dynamics of a piece are increased, loud passages stay loud, soft passages (i.e. below threshold level) become even softer. Dynamic expansion is often used for voice-overs that have a high distortion level. Through expansion, the spoken voice is lifted above the noise level, which is in turn, suppressed.
Expansion can also be used to restore dynamic range to old recordings, LPs and cassettes, for transfer to CD.
When using expansion, a certain reserve of level must exist in order to avoid clipping!

Gate: This mode suppresses audio that is below the predefined gate threshold. Audio above the gate level will pass through unaffected. Very soft passages (below the threshold level) are zeroed. Please note that the gate function is also part of all other modes. Using the dedicated Gate mode reduces processor load if you don't want to apply any other dynamics processing at this stage.

Ratio/Threshold
This section controls the ratio and threshold parameters. You can use the sliders to modify current settings, or type the desired settings directly into the boxes. Notice that the visual display (left of the window) changes to reflect the current ratio and threshold settings.

Note: Click on the "Linear" button to quickly reset the ratio parameter to a 1 to 1 setting (Compressor or Expander disabled).

Attack/Release
This section controls the Attack and Release parameters. You can use the sliders to modify current settings, or type the desired settings directly into the numerical data fields.

Gain
Shifts overall output level to compensate level cut

Gate
This section controls the Level parameter (threshold) of the gate. You can use the slider to modify the current setting, or type the desired setting directly into the numerical data field. Notice that the visual display changes to reflect the current gate level (threshold).
Tip: The gate can be used in any mode by setting the threshold with this parameter.

Play/Stop
Click on this button to audition your dynamics processor changes in real time, without leaving the Compressor/Expander/Noise Gate/Limiter window. All tracks in the VIP will play back, allowing you to make dynamics processor

adjustments in the context of the entire production. You can also solo the processed audio by selecting the solo button in the VIP.

OK

Click on this button to accept the current dynamics processor settings, and close the Compressor/Expander/Noise Gate/Limiter window.

Cancel

Click on this button to close the Compressor/Expander/Noise Gate/Limiter window, without accepting changes to the dynamics processor settings.

Create Copy

As with most of the other Effects menu items, this important option determines whether an Undo will be created prior to processing the file. As always, we strongly recommend that you leave this option enabled!

Bypass

Click on (enable) this option to bypass the dynamics processor. This allows you to quickly toggle the dynamics processing on and off, allowing comparisons between the processed and unprocessed audio.

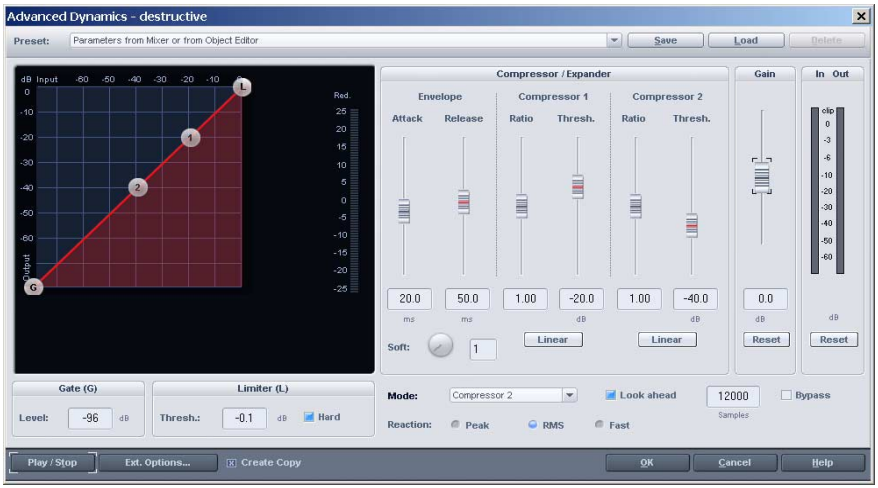
Presets

SAVE: Click on the "Save" button to save the current dynamics processor settings as a preset. A standard Windows "Name File" dialog will appear. Name the preset and click OK, and the preset will be saved.

LOAD: Click on the "Load" button to load a previously saved dynamics processor preset. A standard Windows "Open File" dialog will appear. Select (i.e. click on) the preset you want to load, and click OK. The preset will be loaded.

DELETE: Click on the "Delete" button to delete a previously saved dynamics processor preset. A dialog will appear, asking if you are sure that you want to delete the preset. Click "Yes" to delete the preset. Click "Cancel" to cancel the delete. You can use the preset drop-down list (upper-left side of the window) to quickly load any previously saved dynamics processor presets. Simply click on the menu, and select the desired preset.

Advanced Dynamics



General controls and presets

Hard: When active (i.e. with the switch pressed), the output signal is limited to the defined value. This means that no sample will be louder than the limiter level. Signals are not simply cut off at the limit, as the algorithm attempts to "move" the signals to this limit as gently as possible, without changing the original sound. If the switch is not depressed, the algorithm regulates levels in accordance with the defined transmission characteristic, and behaves like an analogue limiter.

Soft – Knee: This indicates the rounding off of the characteristic lines. Value Range is 1 to 10, where 1.0 means no rounding off. "Rounding off" is practical when the constant change between compressed, and uncompressed, signals is clearly apparent - i.e. the signal level fluctuates around the knee. Knee is used to achieve a soft transition.

Dynamic scope: Visualizes the input and output of the signal during dynamic processing.

Reduction display: Displays the mean level difference between output and input.

Input display: Displays the input level in dB.

Output display: Displays the output level in dB.

Reset: Resets reduction, input and output displays.

Mode: Modes are typical applications of Advanced Dynamics. They constitute default settings, that makes the graphical editing of the characteristics line easier. For this, the number of editable parameters are reduced according to the selected Mode, e.g. by selecting Limiter mode, compressor 1 and 2 are hidden, because they are not needed.

Presets: Saving, loading, deleting: This is where all parameters can be saved, loaded and deleted. The default file extension is *.dy2.
In addition, all Dynamics presets (*.dyn) can be loaded. The parameters are converted to those available in the Advanced Dynamics module. If saved from within the Advanced Dynamics processor, the preset will be saved with the *.dy2 extension.

OK: The parameters are accepted.

Cancel: The parameters will not be accepted.

Help: Opens online Help.

Processing the characteristic lines (graphically)

In order to graphically process the static transmission characteristic, click on one of the nodes in the black grid to the left of the window. A tool tip (which matches the color of the selected node) will appear, indicating the name and value of the parameter. If you move the mouse, with the button depressed, you will change the value. It should be noted that moving a characteristic node may also have an effect on the parameter values of the other nodes, under certain conditions.
To directly position a node in the characteristic field, click on it with the right mouse button. A small dialog field will appear, allowing you to directly enter input and output values.
In order to achieve greater clarity, the different sections are divided into different colors. The color of the tool tips, nodes, and parameter tabs indicate which parameter is being edited.
If you have switched on the Hard Limiter option, and a value lower than 0 dB is selected, a red "LIMITED ZONE" will appear in the characteristic field. The level of samples cannot "enter" this zone, and are kept to a level which is smaller than/equal to the defined limiter level.
In order to hide a node, move the applicable threshold slider to 0 dB. In order to make it visible again, the threshold must be below the limiter threshold.

Processing the characteristic lines (parametrically)

Threshold: This is the input threshold value (in dB). levels above this value cause the respective component to process the audio.

Ratio: Defines the ratio of input level to output level.

Limiter level: level (in dB) determines the maximum level at the output. Please note the HARD setting.

Gate level: level (in dB) determines the minimum level at the input. All signals below this level will have an output signal level of 0 dB.

Make up: Raises the entire static characteristic (in dB), without affecting the limiter and gate settings.
All changes of these parameters have a direct effect on the static transmission characteristic, and on each other. As an example, a gate will render all subordinate level changes ineffective.
All threshold and level settings remain as set. ratio settings can be modified through make-up and Soft. By changing make-up, only the geometric ratio between the output and input level (from the view of the threshold points) is retained. The actual ratio of input to output level is the ratio setting + make-up.

Dynamic parameters
Level measurement methods: The time function determines how the compressor reacts to source material.

PEAK: Allows the dynamic module to react in a more lively fashion, but can tend to be a bit "sharp".

RMS (Root Mean Square): Corresponds to the operating mode of an analogue dynamic module; sounds relatively round and balanced, less lively. The time constant for signal detection (in order to obtain the control signal) is based solely on the setting of the Attack regulator. Both time constants are also evaluated here, for signal smoothing (with signal dynamics manipulation).

Fast: The strength of this process is low processor utilization. This setting can be favorably used to offset peak levels. Use "fast" mode when working at the limits of your CPU, and the dynamics of Objects require minimal adjustment. The maximum level at the output is never higher than the limiter level.

The time constants regulate the sound characteristic substantially, time constants that are too short can lead to distortion effects, while those that are too long can result in "pumping".

Attack: Time constant for the increase of the signal level (in ms). That is, how fast the dynamic processing follows an input level peak.

Release: Time constant for the drop of the signal level (in ms). That is, how fast the dynamics processing reacts on an input level decay.

Preview: If this switch is inactive, the dynamic section does not operate in preview mode. This may result in artifacts (pumping) and over-modulation.

On the flipside, steep attack phases of the wave forms are not smoothed without preview activated – and the sound character therefore tends towards being “brisk and lively”. You should disable preview if trying to emulate the behavior of analogue dynamic modules.

The Dynamic Scope

This scope appears when the Play/Stop button is pressed. A “snow cloud” appears in the characteristic field. Each individual point in the snow cloud represents the difference of output values, above the respective input values (in dB).
The use the Dynamic Scope can be easily explained with the following example: Select preset 0 - “NULL” - and press PLAY. You’ll see that the “snow cloud” is not visible, as all sample levels are on the gray straight line – i.e. all output values are equal to the input values.
If you slide the make-up fader upwards, you can see that the “snow cloud” follows the characteristic. If you select very high time constants, this takes place relatively slowly.
Selecting a relatively small Attack time (10 ms), a long Release time (400 ms) and a severely arched characteristic can move the effect towards “pumping” – give it a try! The “restlessness” of the cloud graphically reflects “pumping”.
By selecting very small time constants, you can visualize the operating mode of a simple harmonic generator.

Tips and hints for the Advanced Dynamics module

In addition to its sonic characteristics, the strength of the Advanced Dynamics effect is in the visually-supported processing of the characteristic. Quick success comes from the selection of one of the supplied presets, and making small adjustments to the characteristic, dependent on the source material. This is very simple, as you can accurately separate the individual signal components with the help of the Dynamic Scope.
If the “snow cloud” moves jerkily, this is probably due to an inappropriate time constant setting. Check and adjust, this setting to suit your tastes/needs. Save your own presets.
In order to couple “analogue” limiting with the hard limiter algorithm, define the Limiter level at the maximum desired sample level. Once set, you can simulate an “analogue” limiter by using an extreme ratio setting (compressor 10.0 or expander 0.1).

Multi -band Dynamics

The multi -band dynamics processor allows you to edit the dynamics, in up to four independent bandwidths (frequency bands). The complete signal, including the control signal, is separated into individual frequency bands, and each band is individually processed by a dynamics processor.
Once processed, the individual bands are reassembled, without any phase shifting, or alteration of the frequency response. Responsible for the high quality process is the FIR filter technique employed during the processing.
If no dynamics processing takes place, and the separated frequency bands are reassembled, the audio material returns to it's original state, without any quality loss.
The biggest advantage of the multi -band compressor (over the standard compressor) is the fact that problem issues such as; obvious "pumping", or other unwanted side-effects are drastically reduced. As an example, a peak in the bass bandwidth would normally "pull down" the rest of the signal, during processing. This issue can be side-stepped with a multi -band processor, set to only compress the peaking bass frequencies. The multi -band compressor allows specific treatment of individual frequency ranges.

Characteristics

The upper-left-side graphic display shows the dynamics reference line. It represents the relationship between the input level (upper label, in dB) and the output level (left-hand label, in dB).
The reference lines always shows the dynamics settings for all frequency bands in the Filter section, each line in its corresponding color. You can move one band on top of the set of curves for a better visibility by selecting one of the

bands edit controls or clicking its according band number (1..4).
Copy to all: Click on this button to Copy the selected band's Dynamics Processing. The settings of this band will also be applied to ALL currently used bands.

Filter Settings

The upper section of the dialog contains the settings for the Filter. The graphic display indicates the approximate frequency responses of the individual bands. Each band has its own colored line. The left axis label displays the absorption (reduction) in dB. The upper numbers show the frequencies in kHz.
This section controls the frequencies of the different bands, and their overlap.

1..3: These sliders adjusts the Cutoff frequency of the Filters. The Cutoff frequency of the fourth Filter is determined automatically by the frequency (and bandwidth, for bandpass Filters) of the adjacent Filter.

You can switch the frequency and dB (amplitude) axes between Lin (linear) and Log (logarithmic) curves. (Log. provides more resolution in lower Ranges, Lin. provides more resolution in higher Ranges.) Use this display to help visualize the different bands.

Gain Enhancement/Reduction

The top-left grid shows the frequency bandwidth, while the display underneath shows the gain reduction being applied to each band.

Setup

Bands
The multi -band Dynamics Processor can function as a broadband, 2-band, 3-band, or 4-band Dynamics Processor. Select the desired number of bands in the list box:

Tip: As you make changes to these settings, notice that the frequency Display updates to reflect the current settings.

Band Separation: Select the amount of the Separation between bands (how much the filters will overlap) in this menu.

Look ahead: If this option is checked, the dynamics section will perform a look-ahead preview. When unchecked, "pumping" or distortion may be introduced. Note that unchecking this option will not automatically "level" steep attack phases of the waveform, which may result in "sharper" audio material. The setting affects all bands.

Link bands: Click on (enable) this option to link all bands of the dynamics processor. Any changes made to a band will be applied to ALL currently used bands.

Bypass dyn.: During audio playback, click on (enable) this option to bypass the currently selected (red) band.

Bypass all: During audio playback, click on (enable) this option to bypass all multi -band Dynamics Processing.

Note: If Link bands is active, it does not guarantee that the settings for all bands are set to the same values. Only parameters that were changed after the Link bands option was activated will match. If all parameters of the bands are to be set to the same values, use the Copy To All option.

Band 1..4

The parameters for the Dynamics section are located to the lower half of the dialog. The controls for each of the four dynamics bands are functionally identical.

Gain

This section controls the make-up gain parameter for the band. This parameter is not available when the selected band is using the Comp Max or Limiter 100 setting.

Ratio: Defines the ratio of input level to output level.

Threshold: This is the input threshold value (in dB). levels above this value cause the respective component to process the audio.

Attack: Time constant for the increase of the signal level (in ms). This is, how fast the dynamic processing follows an input level peak.

Release: Time constant for the drop of the signal level (in ms). This is, how fast the dynamics processing reacts on an input level decay.

Gate

This section controls the level parameter (Threshold) of the Gate. You can use the slider to modify the current setting, or type the desired setting directly into the numerical data field. Notice that the visual display changes to reflect the current Gate level (Threshold).

Mode

Each band's Dynamics Processor can function as a compressor, Limiter, Expander, or Gate. Select the specific type of Dynamics Processing to be applied in the Menu. The functions of the controls (in the Dynamics section) are generally similar to that of the standard dynamics section (Compressor, Limiter, Expander, and Gate). For detailed Information Refer to the "Dynamics" Chapter. In short: The compressor, Expander and Limiter algorithm is similar to that of a "classic" compressor, Expander or Limiter, the Comp Max and Limiter 100 settings will automatically apply maximum make-up gain, by normalizing the

output of the band to 0dB. The Gate is available in all other modes too, as a dedicated mode it saves computing power for that band.

Solo: During audio playback, click on (enable) this option to Solo the currently selected (red) band. This is a very useful feature.

Out (All)
This section controls the output volume (makeup gain) parameter, for the combined (summed) output of all bands. This control is especially useful for getting an idea of what the process sounds like, without adding the volume amplification of the dynamics process. The fader can easily be set to a level that is similar to that of the input signal.

Limiter On (Only in the Mixer Master section)
This option is not available when using the multiband compressor via the Effects menu. When using the multiband compressor in the Mixer Master section, you can switch to, and edit, the Peak Limiter directly from the multiband Dynamics Section. This Peak Limiter provides level limiting when using the multiband Dynamics, increasing the loudness of your mix.

Create copy: As per most of the Effects Menu items, this important option determines whether an Undo will be created prior to processing the file. We recommend that you leave this option enabled!

Play/Stop
Click on this button to audition your Dynamics Processor changes in real-time, without leaving the compressor/Expander/Noise Gate/Limiter window. All tracks in the VIP will play back, allowing you to make Dynamics Processor adjustments - in the context of the entire production. You can also solo the processed audio by selecting the solo button in the VIP.

OK
Click on this button to accept the current Dynamics Processor settings, and close the compressor/Expander/Noise Gate/Limiter window.

Cancel
Click on this button to close the compressor/Expander/Noise Gate/Limiter window, without accepting changes to the Dynamics Processor settings.

Presets
Save: Click on the Save button to save the current multiband Dynamics Processor settings as a preset. A standard Windows "Name File" dialog will appear. Name the preset, and click OK.

Load: Click on the load button to load a previously saved multiband Dynamics Processor preset. A standard Windows "Open File" dialog will appear. Select (click on) the preset you want to load, and click OK.

Delete: Click on the Delete button to delete a previously saved multiband Dynamics Processor preset. A dialog will appear, asking if you're sure that you want to delete the preset. Click Yes to delete the preset. Click Cancel to cancel the Delete.

Tip: You can use the preset drop-down list to quickly load any previously saved multiband Dynamics Processor preset. Simply click on the menu, and select the desired preset.

Strategies to Cope with the Flood of Parameters

A multi -band dynamics processor (naturally) contains many parameters – which can be confusing for inexperienced users. The following strategies may help with editing, using all bands.

If you need to alter all bands, rather than one individual band, the following steps may be useful. The frequency settings are based on the standard configuration:

Step 1: Making Global Preparations for all bands

1. Select the Dynamics Setup that is closest to your desired results. As an example, if you need to add loudness to your audio material, use the compressor Max setting. Use compressor if presence needs to be added, or recorded speech needs become more transparent. This setting will not increase the overall loudness, and is perfect for clarifying muddy recordings, adding a "kicking" bass for pop music etc. The setup only affects the currently selected band (white filter curve). Click on Copy To All to transfer the current setting to the other bands.
2. Activate Link bands. Any changes you perform in the current band are automatically mirrored on the other bands.
3. Change the configuration of all bands, until the audio sounds right to your ears.

Step 2: Fine-tuning of Individual bands

1. Turn off Link bands, and select a band.
2. Activate the Solo mode. Individual bands can be isolated (heard independently) in this mode, making the task of optimizing settings for each band easier. Ideal settings for the higher frequency bands are "small", as the waveforms are shorter.
3. If you can't find satisfying 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.

If you only wish to edit a specific, and critical frequency range, a different approach can be taken:

1. make sure that Link bands is turned off. Activate the Play Solo option, and select the band you wish to edit. If you don't know the exact frequency Range, simply work progressively through some of the bands.

2. Change the split frequencies of the selected band, in order to “filter out” the critical range.

Now you’re ready to perform some dynamics editing. First up, select an appropriate Dynamics Setup.
– To limit critical frequencies, for tasks such as de-essing, Limiter or compressor Setups can be used.
– Use the Bypass Dyn. checkbox to compare the processed and unprocessed bands.
– Turn the Play Solo option off, and compare the original signal with the processed signal. To do so, make use of either; Bypass Dyn. or Bypass All.

A third option is to compare the default presets. Some may provide a great starting point. In most cases, good results can be achieved by starting out with one of the presets, and some judicious tweaking.
The presets, however, will only get you so far. Dynamics processing belongs to the family of algorithms that are entirely dependent on optimal settings, and on the type of audio material being processed.
Different types of audio material will also have a big impact on the type of settings. The original dynamics, the volume level, and the source material are all equally important. Other aspects to consider include; the frequency response of the audio source, and the proposed use of the processed audio.

Comparing at constant volume levels

If the task is to compare the sound differences, without allowing the dynamics processing to alter the volume levels, this method can be used to accomplish the task.
– Use the output All control to adjust the volume until the audio signal level matches that of the bypassed level - i.e. when the Bypass All option is used.
– Compare the levels by repeatedly checking the bypass option, and adjusting the output levels.

If the multi -band compressor is used in the Mixer window’s Master section, the peak meters can be visually used to assist with this task.

Reducing the CPU load

The multiband compressor’s algorithm causes a high CPU load, which is influenced by the settings in the filter section. The CPU load increases with the use of the following settings:
– Low Split frequencies
– Increased number of bands
– Larger settings for the Separation parameter

Should the real-time use (in the Mixer or when using Test for playback) lead to a CPU overload, you may need to attempt matching the results by using settings that are less processor intensive:

1. A high setting for Separation (Normal, High) is only needed in a few special cases. As an example, individual bands may need to be precisely separated from the other bands, for treatment of a very specific frequency area. Typically, this would involve specific bands that are exposed to extreme dynamics manipulations.

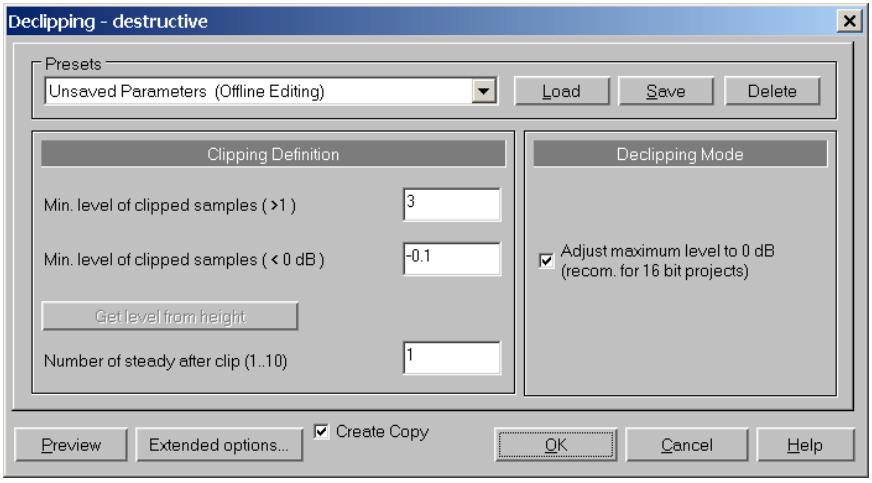
Dynamics treatment of the low bass range often benefits from a high Separation setting. This smoothes the frequency response of the source signal, even for lower settings.

2. The values in the lower split frequencies should not be set to a value that is smaller than necessary.
3. Only use as many bands as are really needed. Three bands are often enough for standard use. As an example, if it's only the bass range that needs to be treated, two bands are sufficient.

Finally, when using the multiband Dynamics Processor, always take care not to overdo the processing. This subtlety is only learnt through experience, but a good general rule to follow is "less is more"....
Careful checking of each band (by using the Play Solo option) will help in this respect.

De-clipping

This unique Samplitude function removes digital or analog clipping. Anybody who records audio has encountered this problem at some point.



Samplitude uses high-grade algorithms to interpolate the "clipped" passages, using the material that precedes/follows the clipped portion as a reference point. Once processed, the total volume level for the material can be further reduced to prevent any additional clipping.
The de-clipping algorithm is especially useful on material that contains obvious but not heavily distorted clipping, such as a piano or voice recording. It does, however, have limits – heavily distorted material cannot generally be repaired, although you may be able to significantly improve the damaged audio.

Clipping Definition

Min Number of clipped samples (> 1): This parameter defines the minimum number of consecutive clipped samples that must occur before Samplitude's de-clipping algorithm will repair the distortion. A value of three works well in most instances.

Min level of clipped samples (< 0dB): This parameter defines the volume level at which Samplitude's de-clipping algorithm assumes that the audio is distorted (In other words a Threshold level, like that found in a compressor). Start with a value of -0.5dB. If clipping persists, reduce this value until the clipped audio is repaired.
A number of sound cards exhibit different clipping behavior, so adjustments to this setting can be an important issue. Some DAT recorders have an analog protection mechanism that prevents the incoming level from reaching the digital maximum signal. For such devices, a setting of –0.5 dB (or lower) makes sense. By entering a value of –6 dB, all samples above half of the digital maximum are considered "distorted", and are recalibrated. Even distorted analog material can be improved with this algorithm.

Note: Once again, it helps if you are patient, and careful, when repairing audio in this manner. If at all possible, take the time to repair the clipped samples sequentially (i.e. one "clipped section" at a time), rather than processing the entire file in one pass.

Get level from height: Click on this button, and the height of the marked Range will be used to define the Min. level of clipped samples (< 0dB) parameter. Maybe you'll have to switch "Fix vertically" in the View menu into "off" state to define vertical ranges. This function is very useful when correcting clipped samples sequentially i.e. one "clip" at a time (see above)

Number of steady samples after clip (1....10): This parameter defines the number of "steady" samples used to determine the end of the distorted segment. In other words, a number of samples that follow the distorted section, that remain at a constant level. A number of lower-end audio cards (e.g. SoundBlaster, and others) produce chaotic values during a clipped segment. For such devices, the only way Samplitude can determine the endpoint of the distorted segment is to increase this parameter value to a setting between 5 and 10. If your audio card is less erratic when playing a clipped segment, you can reduce this parameter to a value of 1.

De-clipping Mode

Adjust maximum level to 0dB – (Recommended for 16 bit recordings, or when in 16 Bit Integer Internal Precision mode) Click on (enable) this option to Normalize the audio, after the de-clipping process. This is recommended if you work with 16-bit material, as there would be no "volume reserve" (headroom) available for the correction of the distortion. When working with

floating point material (24-bit Wave Project), this option can be disregarded, given that the floating point format offers an unlimited dynamic range.

Load Setup
Click on this button to load a previously saved de-clipping preset.

Save Setup
Click on this button to save the current de-clipping parameters as a preset.

Create Copy
As per most of the other Effects menu items, this important option determines whether or not an Undo will be created, prior to processing the file. As always, we strongly recommend that you leave this option enabled!

Preview
Click on this button to audition a (processed) preview of your current de-clipping settings. The duration of the preview is determined by the preview Time setting in the Options > System (Shortcut -Y) menu.

OK
Click on this button to write the current de-clipping settings to the audio file, and close the de-clipping window.

Cancel
Click on this button to close the de-clipping window, without applying any processing.

Remove DC Offset

Choose this option to remove DC-offset from a Wave Project, or selected Objects in the VIP window. DC-offset occurs when the audio waveform is not (vertically) centered at the Zero amplitude Line. In other words, the waveform is centered either above or below the Zero amplitude Line. DC-offset can cause clicks, glitches or other inaccuracies, and should always be removed before applying processing.

Tip: Some audio cards are responsible for the DC-offset while recording, and others are not. To check for DC-offset, Zoom-In on a waveform, and see if it is (vertically) centered at the Zero amplitude Line. If the waveform is not centered at the Zero amplitude Line, use the Remove DC-offset option to fix the problem.

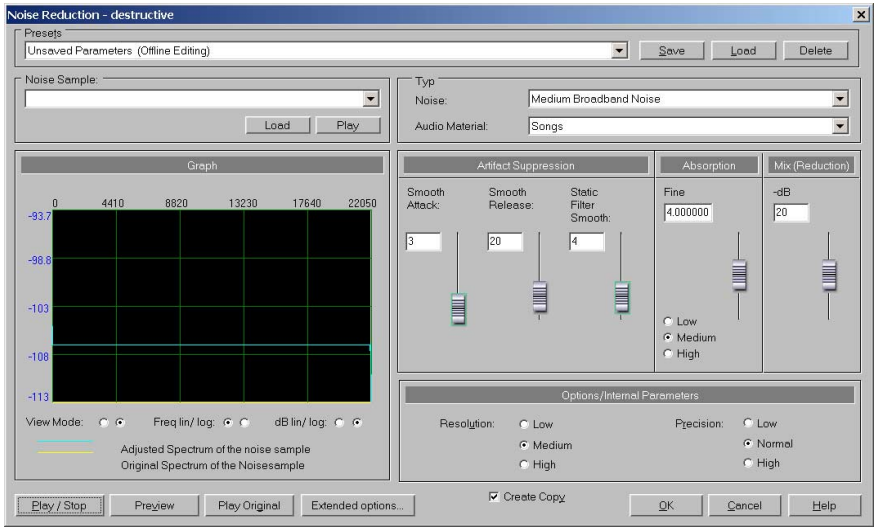
The effect performs in two stages, first it analyses the audio data and then it removes the DC offset. You can set a minimum dB threshold. If the analyzed DC offset is below this value, no removal takes place. This will become useful in batch processing to spare calculation time.

Get noise sample

Use this to get a sample of the noise used in the noise reduction function.
For general information on noise reduction read the according manual chapter.
This function is the first step in the noise reduction process.
The noise reduction function can be used to effectively remove annoying noisy material from the audio, with minimal discoloration or other artifacts. In order to achieve this, the algorithm first needs a sample (noise print) of the noise that you wish to remove. This is most effective with constant noises such as; a ground loop, air conditioner hum, tape hiss or feedback.
Before using noise reduction, you must first make a noise print (of the noise you wish to remove).
To do so, mark a Range in the Wave Project or selected Object that contains ONLY the noise. Then, select this option to take the actual “Noise print”.
Note: If the “Noise print” contains audio (as well as the noise), it will also be removed. So....be sure that the “Noise print” ONLY contains noise!
The duration of the noise print should not be too short – Samplitude will display a dialog if a longer sample is required.

Noise reduction

(Please refer to previous Item – get *noise sample* before proceeding)
This option opens the noise reduction window. This window allows you to apply graphic “Noise print” style noise reduction to Wave Project, or selected Objects in the VIP window.



Important note: You must first use the get noise sample option to take a “Noise print” (of the noise to be removed) before using noise reduction.(see previous Item)

Here is a quick overview of the steps involved:

1. Select a noise sample as explained above.
2. Next, select the section in the Wave Project (or select the VIP Object) that requires the removal of the noise signal.
3. Open the noise reduction dialog via the Effects menu.
4. In the noise sample box, select the noise sample setting. This file contains the noise print.
5. You may wish to preview the changes before committing them to the audio material. To do so, click on the preview button.
6. make all desired changes in the dialog, or load a preset and click on OK to process the sample.

Noise sample

The "Noise print" created with the get noise sample option will automatically be displayed (selected) in this drop-down list.

Tip: Any loaded Wave Project can be selected from this drop-down list, and used as a "Noise print". As an example, say you've recorded several guitar tracks through an amplifier that generates a significant amount of 60 cycle hum.
Rather than taking a "Noise print" for each individual Track, you can mark a Range in one Track (containing the hum - in isolation) and copy that "Noise print" to a new sound file (Wave Project). You can then use this sound file ("Noise print") to remove 60 cycle hum from ALL guitar tracks recorded through the noisy amplifier. In other words, you can create an archive of commonly used "Noise prints" to speed up repetitive noise reduction tasks.

Play

Click on this button to listen to the "Noise print". Check for evidence of audio you may not want to remove from the file. If you can hear audio you don't want removed, create a new noise print.

Draw Mode

In the first mode, the frequency response is displayed as a solid curve. In the second mode, the frequency response is displayed as a line.

Freq lin / log

Linear display of frequency provides a more detailed view of the higher end. Logarithmic display of frequency provides a more detailed view of the lower end.

dB lin / log

Linear display of amplitude provides a more detailed view of the higher end. Logarithmic display of amplitude provides a more detailed view of the lower end.

Presets

(Type of) noise: To make the noise reduction process quicker/easier, you can select presets that specify the type of noise (to be removed. Samplitude will adjust the noise reduction parameters accordingly.

Medium Tonal noise (e.g.: hum): Use this preset for reducing 60 cycle hum, and other tonal noise.

Hard Tonal noise (e.g.: hum): Use this preset for reducing louder 60 cycle hum, and other louder tonal noise.

Low broadband noise (e.g.: hiss): Use this preset for reducing low-level hiss, and other low-level broadband noise.

Medium broadband noise (e.g.: hiss): Use this preset for reducing louder hiss, and other louder broadband noise.

Hard broadband noise (e.g.: hiss): Use this preset for reducing severe hiss, and other severe broadband noise.

Gramophone 1 Medium (e.g.: vinyl): Use this preset for reducing vinyl noise.

Gramophone 2 Hard (e.g.: vinyl): Use this preset for reducing louder vinyl noise.

(Type of) Audio Material: To make the noise reduction process quicker/easier, you can also select presets that specify the type of audio (from which the noise will be removed). Samplitude will adjust the noise reduction parameters accordingly.

Speaking Voice: Use this preset when applying noise reduction to Speech.

Singing Voice: Use this preset when applying noise reduction to Vocals.

Fast Beats / percussion: Use this preset when applying noise reduction to percussion or other short "choppy" audio.

Songs: Use this preset when applying noise reduction to complete mixes.

Orchestral Instruments: Use this preset when applying noise reduction to individual instruments.

Tip: Use these presets. They'll enable you quickly "dial in" noise reduction settings that produce the desired result, and minimize potential side-effects.

Algorithm's Internal Parameters

Resolution: This parameter allows you to set the resolution (Low, Medium, or High) used by the noise reduction algorithm. The higher the resolution, the longer the algorithm will take to process. You may wish to experiment with this parameter because a higher resolution does not always guarantee the best results. As an example, Low resolution often works best when applying noise reduction to short percussive sounds, and speech.

Precision: This parameter allows you to choose the quality level that will be used to for noise reduction. Low, Normal, and High are the three choices, with High offering the best quality, but taking the longest to process.

Precision Real-time: This parameter allows you to choose the quality level used for real-time previewing of noise reduction. Test, Low, Normal, and High are the four choices, with High offering the best quality, but requiring the most processing (CPU) resources.

Tip: If your machine isn't fast enough to provide a real-time preview at the High setting, try one of the lower settings. The quality of the preview will be slightly lower, but this won't affect the quality of the actual noise reduction process on your audio file.

Absorption

This parameter determines the intensity (Low, Medium, or High) of the noise reduction algorithm. You can use the Fader to fine-tune the setting.
The low, medium and high settings determine the effectiveness of the algorithm. The fader allows fine-tuning of the setting, which always operates in the selected setting range. This parameter sets the level of the noise print sample absorption (or amplification). This is visible in the graphic display. The yellow curve indicates the original spectrum of the noise print sample, and the blue curve represents the corrected spectrum. The latter is used to remove the noise from the audio.

Tip: In most cases, you will want to adjust the Absorption parameter to be just aggressive enough to reduce the noise. Extreme settings are likely to cause artifacts.

Mix (Reduction)

This parameter determines the amount (in dB units) that the noise is reduced.

Tip: Quite often it is actually beneficial to leave a little noise in the audio. Removing all of the noise is usually impossible without damaging the remaining audio. Such damage can include artifacts such as; strange flanging sounds and whistles. Think of the process in terms of reduction, not 100% elimination. As always, be patient, try a number of settings, and select the ones that deliver optimal results.

Artifact Suppression

When trying to reduce high levels of noise, artifacts can often occur (see above). This isn't a limitation of Samplitude, but rather a universal by-product of current noise reduction techniques. If you encounter artifacts when applying noise reduction, the following three parameters may prove useful (in addition to reducing settings such as Absorption and Mix).

Smooth Attack: This parameter controls the Attack when applying noise reduction. A higher value will reduce artifacts, but alters the response time of the audio. Generally speaking, this parameter is most effective for diminishing artifacts in musical tracks.

Smooth Release: This parameter controls the Release when applying noise reduction. A higher value will reduce artifacts, but alters the decay of the audio. Generally speaking, this parameter is most effective for diminishing artifacts in percussive (instruments with short decay) tracks.

Static Filter Smooth: This parameter "levels" the Spectrum of the noise sample. (To better understand what this parameter does, look at the blue curve in the graphic display.) This "leveled" Spectrum is then used to calculate the noise reduction. Generally speaking, this parameter is most effective for diminishing artifacts in vocal, or other "smooth", tracks.

Create Copy

As per most of the other Effects menu items, this important option determines whether or not an Undo will be created, prior to processing the file. As always, we strongly recommend that you leave this option enabled!

Play/Stop

Click on this button to audition the noise reduction in stereo.

Preview

Click on this button to hear a calculated (processed) preview of the noise reduction. This option should be used if your machine isn't fast enough to provide a real-time preview. The duration of the preview is determined by the preview Time setting in the Options > System (Shortcut -Y) menu.

Play Orig

When using the Review functions, click on this button to play the original (dry) audio. This allows you to quickly compare processed vs. unprocessed audio.

OK

Click on this button to write the current noise reduction to the file, and close the noise reduction window.

Cancel
Click on this button to close the noise reduction window, without applying any processing.

Load Setup
Click on this button to load a previously saved set of noise reduction parameters.

Save Setup
Click on this button to save the current noise reduction settings.

To use noise reduction:
1. If noise reduction is to be applied directly to an HD Wav or RAP file, ensure that it (Wave Project) is the current active window. If noise Reduction is to be applied to an Object in the VIP window, ensure that the Object is selected. (Click on the Object in the lower half of the Track to select it.)
2. For this example, we will remove 50/60 cycle hum from a guitar Track. First, mark a Range (in he Wave Project or selected Object) that ONLY contains 50/60 cycle hum.

Tip: Click and drag in the Wave Project window to mark a Range. Click and drag (starting in the upper half of the Track) to mark a Range in the VIP window.

3. Select *Effects* > get *noise sample to obtain* a „Noise print" of the 50/60 cycle hum.
4. If noise reduction is to be applied directly to an Wave Project file, mark the desired Range in the Wave Project window. (Click and drag in the Wave Project window to mark a Range.) If noise reduction is to be applied to a selected Object in the VIP window, proceed to step 5.
4. Open the noise reduction window from the Destructive Effects menu.
5. Under the preset section, select *Medium Tonal noise* for the Noise parameter, and select either Songs or Orchestral Instruments for the Audio Material parameter.
6. Click on one of the *Play* button to hear a real-time audition of the noise reduction. If your machine isn't fast enough to keep up with the load of a real-time preview, click on the preview button and Samplitude will process a calculated preview.
7. If you are satisfied with the results, click on *OK* to write the noise reduction to the audio file. If you are not satisfied with the results, adjust the resolution, Absorption, and Artifact Suppression parameters, and repeat steps 6 and 7.

Noise reduction Tips
To identify optimal settings please follow the steps below:
1. While familiarizing yourself with the noise reduction processor, always begin by selecting preset parameters which are appropriate for the type of audio material that needs correction, and the type of noise.
2. Experiment with the Absorption parameter. Never over-do this setting! Your audio could lose all of it's ambience if this setting is too high.

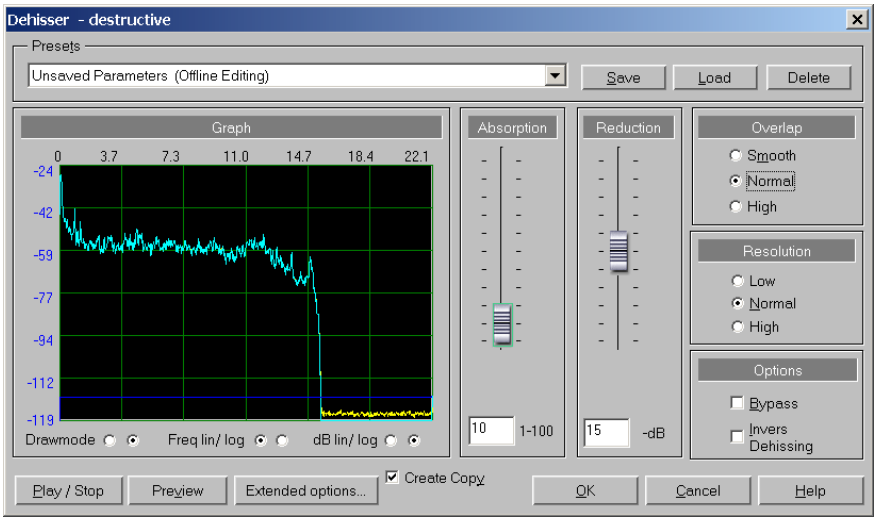
3. To reduce potential artifacts, increase the Attack and/or Static Smoothing values. The selected method depends on the audio material that needs treatment.
4. Always be aware that the Artifact Suppression parameters are affected by the Absorption parameter, and vice versa. Always search for the optimal relationship - which balances artifacts against removed noise.
5. It is good practice to check files for DC Offset and remove it (if there) prior to using noise reduction. (Effects > Remove DC offset)
6. Discoloration of the original audio material can be reduced by repeated use (i.e. multiple passes) of the noise reduction algorithm. A low Absorption level should be chosen. Once the first attempt has been executed, the remaining noise signal in very low volume sections should be used as the noise print for the next attempt.
7. High level broadband noise can be difficult to remove. It is therefore suggested that you treat the material - prior to using noise reduction - with a filter. The Equalizer can be used to cut frequencies that are not used by the original (desired) audio. As an example, a female voice doesn’t use frequencies below 200 Hz, and above 8000 Hz, so these can safely be reduced (but ONLY if necessary!).

   For instruments - everything below the lowest note can be reduced. As an example, if the lowest note played is an A=440Hz then everything below 440 Hz can be cut. This may result in the loss of some harmonics, but audio repair always involves some trade-offs. The FFT Spectrum Analyze Filter (*Effects > FFT Filter/Analyzer*) is best suited for this task.
8. Whether or not artifacts are audible, is particularly dependent on the volume of the end result, and the frequency response of the audio playback system. If you know where the audio will be reproduced (movie theatre, TV, radio, etc.) it may be helpful to repair the audio in similar conditions.
9. For sound designers: The algorithm can be used to achieve interesting discoloration of the original material, especially if the noise sample contains material other than that of the noise signals.
10. The algorithm is useful for the removal of incorrectly played notes, if the notes/tones contain a spectrum with overtones. To accomplish this, a sample that only contains the wrong note is required. This wrong note can then be captured as a noise sample.

Once again, the “less is more” approach applies to effective noise reduction. As the audio engineer, the noise will be far more noticeable to you than most listeners. On the flipside, most listeners quickly notice processing artifacts such as flanging/whistling, particularly on vocals. As such, a little remaining noise is preferable to having weird artifacts in the audio.

De-hissing

This option opens the de-hisser window, allowing you to apply de-hissing to Hard Disk (HD Wave) and RAM Projects (RAP), or selected Objects in the VIP window.
The de-hisser complements the noise reduction DSP. It can be used to remove constant low-level white noise. This is typically found in material recorded through low quality pre-amps, or low quality analog sound cards.
The de-hisser does not require a noise-print sample, as is necessary with the noise reduction DSP. This makes the de-hisser ideal for real-time use in the Mixer window's Master section.
The dialog of the de-hisser (when using it in real-time) differs from the dialog used for off-line processing ('Effects' menu).
The de-hisser was combined with a simple FFT (Fast Fourier Transform) filter, for use in the Master section of the Mixer window. As the de-hisser algorithms make use of FFT functions, the audio is only processed once, for both DSP steps.
When using the off-line version of the de-hisser (*Effects > de-hissing*), additional control options are available.



Graphic Display Curves

Three different Curves (each a separate color) are displayed by the graph.

Light Blue: The light blue Curve represents the frequency response that results from the de-hissing process.

Yellow: The yellow Curve represents the original frequency response (before applying de-hissing).

Dark Blue: The dark blue Curve (flat line) represents the "threshold" of the de-hissing algorithm (defined by the Absorption parameter value).

Draw mode
In the first mode, the frequency response is displayed as a solid Curve. In the second mode, the frequency response is displayed as a line.

Freq. lin / log
Linear display of frequency provides a more detailed view of the higher end. Logarithmic display of frequency provides a more detailed view of the lower end.

dB lin / log
Linear display of amplitude provides a more detailed view of the higher end. Logarithmic display of amplitude provides a more detailed view of the lower end.

Absorption
This parameter determines the intensity of the de-hissing algorithm. You can use the fader to modify the current setting, or you may type the desired setting directly into the box.
This parameter determines the threshold between the noise and audio signals. Finding the correct settings for this parameter is of utmost importance. Unlike noise reduction, where a useful Absorption value is automatically entered, the value in the de-hisser dialog needs to be manually defined, as it is highly dependent on the type of audio material.
Low settings can lead to incomplete removal of noise, or may introduce artifacts. High settings may discolor the audio material. The louder the volume of the noise print, the more attention you need to pay to the settings in this dialog

Tip: In most cases, you'll want to adjust the Absorption parameter to be just aggressive enough to reduce the hiss. Extreme settings will, in all likelihood, cause artifacts.

Reduction
This parameter determines the amount (in dB units) of hiss reduction. You can use the fader to modify the current setting, or you may type the desired setting directly into the box.
The highest setting is –30 dB. If the reduction value is low, side effects from the algorithm (artifacts) remain low. These artifacts can be introduced in situations where the noise print is relatively loud, or the Absorption settings are not optimal.
Extremely low volume sections (such as the Fade Out of an instrument sound) are often critical, as the actual noise print level can exceed that of the audio signal. These situations may necessitate the use of smaller values for the reduction parameter (-10 to –15 dB). This may reduce the introduction of artifacts, and could prove to be a good compromise.

Resolution
This parameter allows you to set the resolution (Low, Normal, or High) used by the de-hissing algorithm. The higher the resolution, the longer the algorithm will take to process. You may wish to experiment with this parameter, as a higher resolution does not always guarantee the best results. As an example, Low resolution often works best when applying noise reduction to short percussive sounds, and speech.

Overlap
This parameter selects the type of "overlap" used by the de-hissing algorithm (Smooth, Normal or High).

Smooth: Generally speaking, this setting is most effective when applying de-hissing to musical tracks (or audio with slower attack).

Normal: Generally speaking, this setting is most effective when applying de-hissing to percussive (instruments with short decay) tracks.

High: Generally speaking, this setting is most effective when applying de-hissing to mixed (multiple instruments) tracks.

Bypass
During audio playback, click on (enable) this option to bypass the de-hisser.

Inverse de-hissing
When using the de-hisser, click on (enable) this option to limit what you hear to the hiss that is being removed. This is a great way of ensuring that you aren't causing side effects by using the de-hisser. If you hear any significant amount of signal (besides the hiss), you should probably readjust the Absorption and reduction parameters.

Note: You will find this option to be an essential part of your de-hissing routine.

Create Copy
As per most of the other Effects menu items, this important option determines whether or not an Undo will be created, prior to processing the file. As always, we strongly recommend that you leave this option enabled!
When this box is checked, Samplitude will create a copy of the selected Range into the same audio file as the original material. This allows you to compare your results with the original material. Please consider the length of the selected Range, and the available hard disk space or RAM memory.

Play/Stop
Click on this button to hear a real-time Stereo audition of the de-hissing process.

Preview
Click on this button to hear a calculated (processed) preview of the de-hissing. This option should be used if your machine isn't fast enough to provide a real-time audition.

Tip: You can determine the length of the preview by pressing the Y key (to open the System window – Program options) and entering the desired number of seconds in the preview Time parameter box (center of the window).

Bypass.
Click on this button to play the original (dry) audio, when using the Playback functions. This allows you to quickly compare processed vs. unprocessed audio.

Save Setup
Click on this button to save a set of de-hisser parameters.

Load Setup
Click on this button to load a previously saved set of de-hisser parameters.

OK
Click on this button to apply de-hissing, and to close the de-hisser window.

Cancel
Click on this button to close the de-hisser window, without applying any processing.

To apply de-hissing:
1. If de-hissing is to be applied directly to an Wave Project file, ensure that it (Wave Project) is the currently active window. If de-hissing is to be applied to an Object in the VIP window, ensure that the Object is selected. (Click on the Object in the lower half of the Track to Select it.)
2. If de-hissing is to be applied directly to an Wave Project file, mark a Range in the desired Wave Project window. (Click and drag in the Wave Project window to mark a Range.) If de-hissing is to be applied to a selected Object in the VIP window, proceed to step 3.
3. Select *Effects* > de-*hissing*, and the de-hisser window will open.
4. Enter a value of 40 for the Absorption parameter.
5. Enter a value of 12 for the reduction parameter.
6. Click on the *Play/Stop* button and Samplitude will process a calculated preview.
7. If you are satisfied with the results, click on *OK* to write the de-hissing process to the audio file.

If you are not satisfied with the results, adjust the Absorption parameter so that it is just high enough to reduce the hiss (without producing artifacts). You should also readjust the reduction parameter. You may also wish to

experiment with the resolution and Overlap parameters. Once your setting have been made, repeat steps 6 and 7.

Tip: As previously mentioned, it's a good idea to click on (enable) the inverse de-hissing option (to verify that you aren't removing too much of the desired signal) before applying your de-hisser settings.

Suggested steps for finding the optimal Absorption parameter setting:

1. For preview purposes, locate a "critical" section in your audio material. Critical sections comprise portions of your audio material that contain low level music or speech. Other such sections are those that contain a relatively high level of noise when compared to the audio signal, or audio that is similar to particular types of noise signals. As an example, the "breath" sound of a saxophone tone may have similar characteristics to a noise signal. Another example is the sibilance of spoken words. It is easier for the algorithm to distinguish sounds with overtones, or consonants, from noise signals.
2. Set the reduction setting to the highest value (-30 dB).
3. Slowly increase the value of the Absorption setting. This can be performed during playback of the real-time preview.

Adjustments to the Absorption setting causes the audio to go through four possible stages:
Stage 1: If the value is small, no noise is removed.
Stage 2: The noise is partially removed. Dependent on the level of the noise signal, a small number of artifacts may be introduced, if the noise is not completely removed.
Stage 3: The noise is completely removed.
Stage 4: If the value is high, the algorithm not only removes the noise signal, but also portions of the audio signal. (Use the inverse option to verify the material that is removed. If portions of the audio signal are included in the audible signal, the Absorption value is too high.) The audio material loses definition and sounds dull. This is due to the removal of ambient characteristics. The task is to reach Stage 3.

Tip: When auditioning the changes, it often helps to use a high monitoring level. It may also help to tweak the high EQ on your mixing console or amp, but be sure to return it to it's "neutral" position before finally accepting the setting! We also recommend the use of a system with a good signal-to-noise ratio.

4. If the noise signal can not be removed without discoloring the music or speech, reduce the reduction value until a compromise is reached between noise reduction/minimal discoloration.

If you destructively reduce the noise of individual audio tracks in a multi-track VIP, the de-hisser settings become less critical than when using the de-hisser, in the Mixer Master section, for the complete mix. For higher

noise levels, you should therefore use the “Effects” menu version of the de-hisser, and remove the noise from the tracks individually.

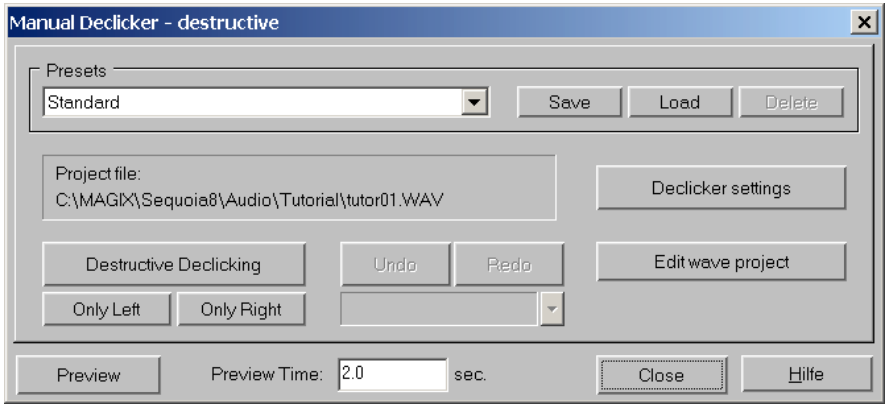
Note: When working in multi-track projects, if the hiss is limited to one or two tracks, you will almost certainly achieve better results by de-hissing the individual tracks, as opposed to inserting the de-hisser in the Mix Master channel in real-time. It is far better to only change the tracks affected by hiss.

Once again, the “less is more” approach is often the most suitable for effective de-hissing. As the audio engineer, the noise will be far more noticeable to you, than to most listeners. Most listeners will, however, quickly notice processing artifacts such as flanging/whistling, particularly on vocals. As such, a small amount of noise is preferable to having weird artifacts in the audio, in most cases.

Note: When using the real-time de-hisser in the Mixer Master channel, Samplitude currently requires a VIP Buffer setting of at least 16000. This can be set in the Options > System dialog (Shortcut – Y). If the buffer is not set to this value (or higher), the de-hisser will not function. In fact, it will refuse to display.

Manual de-clicker

This dialog enables the destructive de-clicking of individual objects. It contains the de-clicker, a button for direct Wave Editing as well as Algorithmix reNOVAtor support.



Settings can be loaded and saved. Self-created settings saved in the FX_presets-directory are displayed in the preset list.

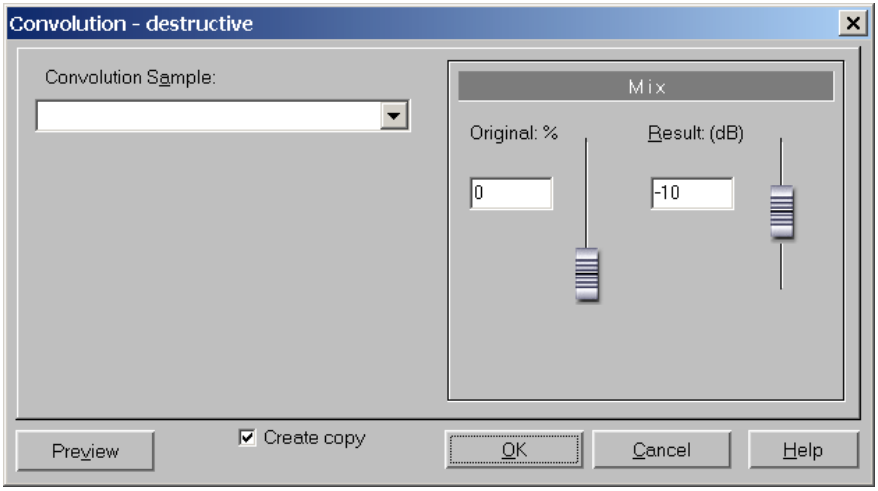
Spectral Editor - Algorithmix reNOVAtor

Samplitude provides a demo version of the Algorythmix reNOVAtor. This high-quality effect is most suitable for eliminating coughs, whispers or other interference.

By marking the interference in the visible spectrum, the interference is "erased" with the help of a sophisticated Algorithm. The "Algorythmix reNOVAtor" must also be unlocked.
Read more about applying this effect in the Algorithmix reNOVAtor Quick Start manual.

Convolution

Choose this option to open the Convolution window. This window allows you to (destructively) apply the "characteristics" of an audio segment to other Wave Project, or selected Objects in the VIP window. It might help to think of this processor as a "simplified" version of the Room Simulator (they are both Convolution Filters).
When convolving two samples, the original sample will be amalgamated (by a complicated filtration method) with the convolution sample. Common overtones will be amplified, different overtones will be reduced, and a number of ambient characteristics will be transferred.



Note: A similar technique is available in the FFT Analyze Filter - if a sample is filtered with the saved frequency response from another sample. This method, however, only considers the frequency responses of the two samples.

Convolution can result in innovative reverb, delay, filter, or morph-effects between the two samples. Unexpected results can take your Project in new directions!
The basics of this method are the same as those of the Room Simulator. In this case, an exact mathematical convolution function is used, whereas the convolution function is optimized for calculating reverb responses in the Room Simulator. The maximum length of the Convolution sample is double that of the Room Simulator.

Convolution sample

Use this Drop-Down List to select the audio segment that you wish to use for the "characteristics" to be applied to the current Wave Project or selected Object in the VIP window. When using files longer than 1048576 samples, only the first 1048576 samples will be used.

Important note: The segment of audio (Convolution sample) must first be loaded into Samplitude (as an Wave Project) before it can be selected from this list.

Mix

Original (%): This parameter allows you to adjust the level of the original (dry) audio. The data can be entered numerically (via keyboard) in the adjacent data field.

Result (dB): This parameter allows you to adjust the level of the resultant (wet) audio produced by the Convolution process. The data can be entered numerically in the adjacent data field. Values above -10 dB can sometimes produce distortion or clipping, so it may be advisable to use small values initially.

Create Copy

As per most of the other Effects menu items, this important option determines whether or not an Undo will be created, prior to processing the file. As always, we strongly recommend that you leave this option enabled!

Save Setup

Click on this button to save the current Convolution parameters as a preset.

Load Setup

Click on this button to load a previously saved Convolution preset.

Preview

Click on this button to audition your current Convolution settings. The duration of the preview is determined by the preview Time setting in the Options > System (Shortcut -Y) menu.

OK

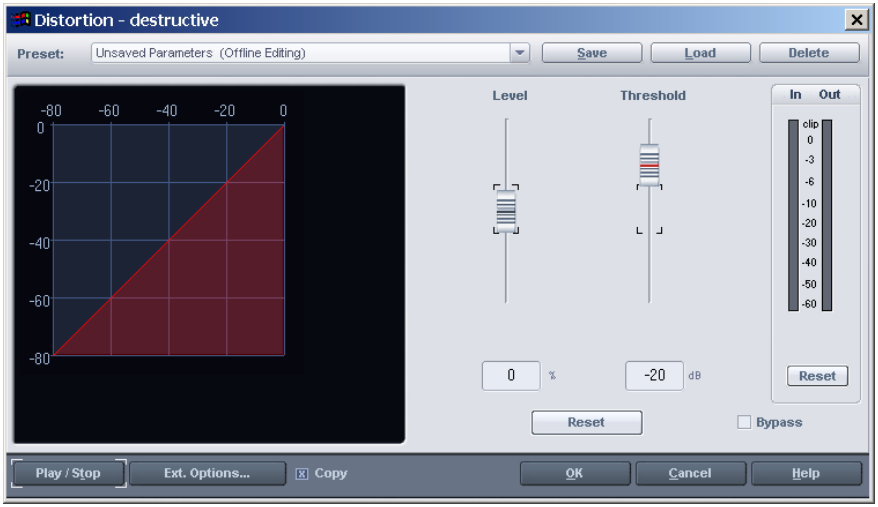
Click on this button to write the current Convolution settings to the audio file, and close the Convolution window.

Cancel

Closes the Convolution window, without applying any processing.

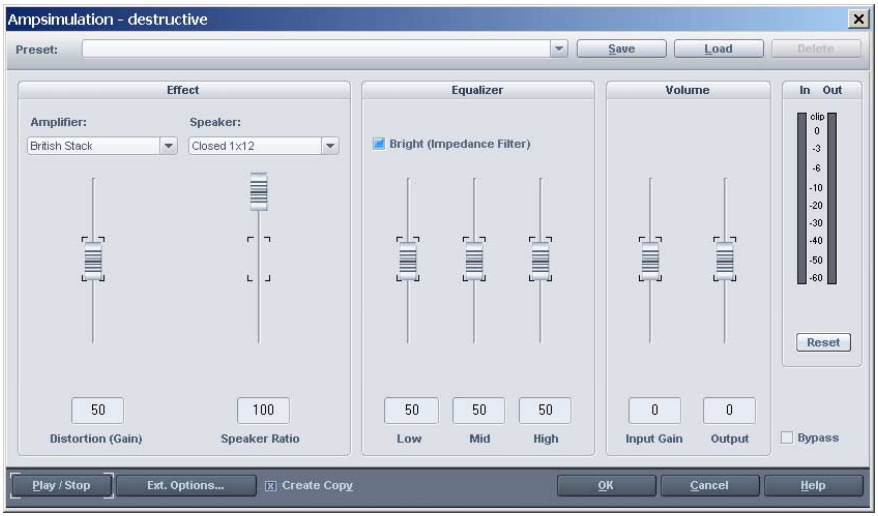
Distortion

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 (i.e. Threshold at –40 dB) can be created (Overdrive). A harsh, digital sounding distortion can be created with a Threshold setting of 0 dB. The ratio setting determines the strength of the distortion.



Amp simulation

The Amp simulator is a real-time effect that simulates tube amp sound properties. It is especially suitable for guitar sounds, but also for the organ or recordings of vocals.



Amp Model: Here you can select from a range of amplifiers whose sound can be simulated.

Speaker Model: Here you can select from a range of speakers whose sound can be simulated.

EQ-Section: A 3 band equalizer for setting the bass, mid-tones and treble.

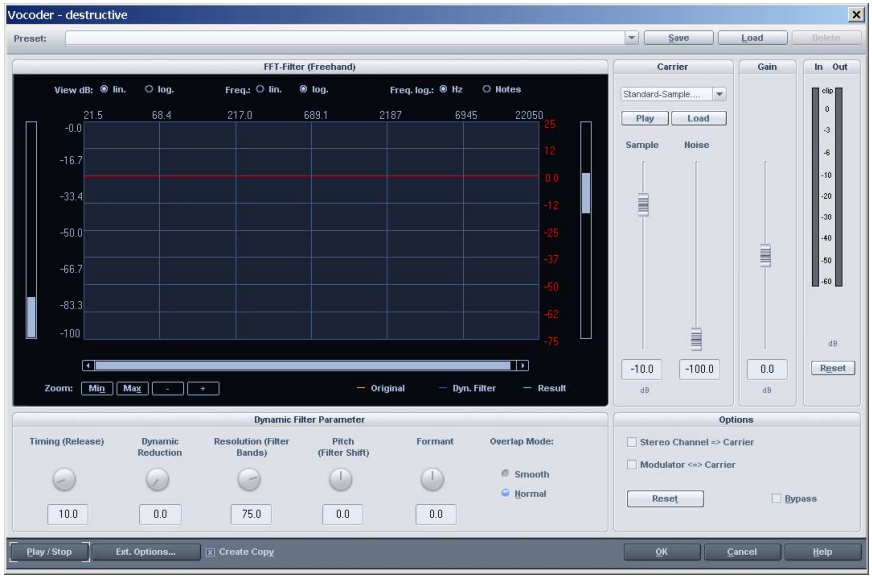
Bright: The Bright button is for a particularly sharp sound display, that can be jarring when used with the following automatic controllers:

Distortion: Creates distortion in typical analog amplifier fashion.

Volume: Controls the entire volume of the amplifier module.

Vocoder

The Vocoder in brief: Carrier material (e.g. a string instrument pad or a Synth chord) are affected by a modulator (e.g. language or singing, or even drum loops) to give the impression that the material is "speaking" or "singing". This is done 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 have only one input, the carrier is either "won" from within the effect, which can then be mixed precisely with white noise or any WAV file. Or you can use the Vocoder as a mono effect whereby one input channel is used as carrier and the other one as the modulator.
There is a Real-time FFT filter within the Vocoder for editing Vocoder signals.

Vocoder control elements

Filter curves: The yellow line represents the frequency properties of the modulator. The light blue line represents the frequency properties of the carrier. The red line is a freely plottable FFT Filter. Any frequency property can thus be drawn in to optimize the results of the Vocoder. The blue line is the resulting filter curve of the Vocoder. For more about zooming and filter graph display options, please read the FFT filter help files.

Reset filter: resets the red filter curve to its initial state.

Dynamic filter parameters

Response time (Release): Affects the speed of adjusting the dynamic filter to the modulator spectrum. The greater the value, the slower the Vocoder follows the modulator and softer the sound changes in the carrier. Give the parameters a low value for better results.

Dynamic reduction: Affects the dynamics of the modulator signal for reducing the modulation depth of the dynamic filter.
This prevents two modulation side effects: Firstly, modulator signal volume adjustment is transferred to the output signal in a less complex form. This improves the Vocoder voice in the mix. Secondly, low-level portions of the modulator signal are ignored so as to avoid a modulation of the carrier by breathing sounds or signal interference.
This parameter is useful only if the modulator signal is fully controlled.
Instead of using dynamic reduction within the Vocoder, you can adjust the modulator's signal dynamics (or the results) with the dynamic tools integrated in the program.

Resolution (Number of filter bands): Dynamic filter resolution (approximately dependant upon the number of filter bands). The best results occur with medium to high resolution.

Pitch (Filter shift): The dynamic filter of the Vocoder shifts the frequency up or down to create certain pitch effects. For the best results, don't change these parameter.

Formant: Tosses or stretches the dynamic filter curves to manipulate the formants, thus changing the characteristics of the Vocoder voice.

Smooth/normal overlapping: Internal parameter that changes the overlapping time window (which calculates the modulator signal spectrum). In the "Smooth" mode, the Vocoder sounds are softer and more tuneful, but quality may suffer.

Carrier signals

Sample selection list: Carrier samples are selected here. All opened Wave Projects are listed along with some special carrier samples loaded automatically from the Vocoder directory. Carrier samples prefer materials with even frequencies e.g. orchestral chords, broad synthesizer surfaces, wind noises etc.

Sample (dB): Sets the number of carrier samples.

Noise(dB): Automatic controller that mixes white noise into the carrier. This is above all useful if the carrier material can't be modulated well or sounds too uneven. Whispering voices can also be produced in this fashion. It's also good for filling in drum loops between beats that are too far apart.

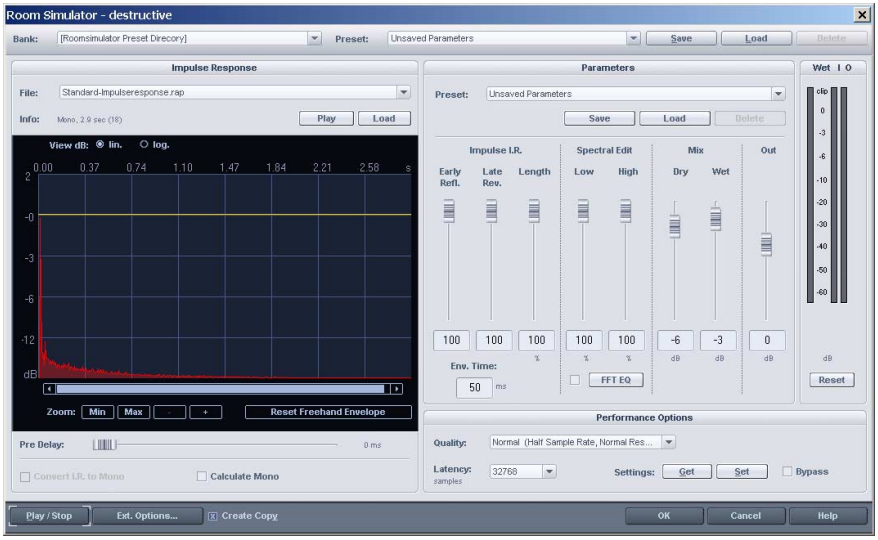
Modulator/Exchange carriers: Exchanges modulator and carrier signals. This is particularly useful if the "stereo channel as carrier" option is used.

Stereo channel as carrier: If this option is active, as carrier signal no more sample from the selection list used, but a stereo channel of the input signal, the other channel serves as modulator. The advantage is the resulting accurate synchronization between carrier and modulator signal - the result is always independent of the play start.

Here are some short instructions:
Create 2 tracks which each include a modulator and carrier. Then create a submix track including the Vocoder as an effect with the "Stereo channel as carrier" option. Route the outputs of the modulator and carrier tracks ("Track" menu) to this submix track and then set the modulator pan to the left and the carrier pan to the right. Instead of using audio files you can also use VST instruments, or you can combine audio files with VST instruments.

Volume: Adjusts the Vocoder output level.

Room Simulator



General Information

This function simulates the reverb patterns of any room reverb signal. The process calculates the impulse response of the reverb and applies it to the audio material in the selected range of the object in the VIP track. A room impulse response is the natural decay of a sharp, high-impact like sound such as a gunshot or similar. It contains all necessary information to exactly replicate the room reverb. The impulse response can be influenced with the dialog parameters. The reverb characteristics are accessible for easy editing. In addition, the reverb character can be determined by selecting the type of impulse response. A graphic display of the impulse response and additional fade-curves allow you to get a visual overview of the impulse response manipulation via the Room Simulator dialog. In Samplitude professional, the Room Simulator is available as a real-time effect in objects, tracks, AUX busses and master channels

General operating elements

Bank

For selecting a bank directory into which the impulse response list is transferred.

Presets (Impulse Response and Parameters)

When loading a preset, the Room Simulator loads parameters of the dialog and the impulse response associated with the settings. The impulse response can be any sample that is saved to the hard disk as a WAV, HDP, or RAP file. If the file was moved or deleted, the Room Simulator displays an error message. A special *.imp file is created for standard impulses.

When saving a preset, only the file path to the selected impulse response saved, and not the impulse itself.

Load/Save/Delete presets (Impulse Response and Parameters)
The presets can be saved, loaded, or deleted. The load function is only needed if the preset files are located in a different folder than the "RoomSim" folder. (Presets that are located in this folder are automatically loaded and displayed in the selection list of the dialog.) The presets have the extension *.rms.

File (Impulse response)
Here you can select an impulse response from the "RoomSim" and Bank directories as well as any currently opened Wave Project. If the Wave Project does not contain an impulse response, no reverb effect will be produced.
If you deactivated the "Install Impulse Responses" option when loading the program, the directory remains empty. If you skipped this part of the installation you may always go back and install just this part of the Samplitude installation.

Load (Impulse response)
The impulse response can be any sample that is saved to the hard disk as a WAV, HDP, or RAP file.

Play (Impulse response)
To verify the impulse response, it is played after this button is activated. IMR file impulse responses are only played with 16 bits, even if the impulse response has a 32 bit float format..

Graphics

dB lin
The graphic display will follow the usual sample display based on the linear amplitude scaling applied to the graphic.

dB log
The scaling of the amplitude display follows a logarithmic pattern. This coincides with the way the human ear perceives volume changes.
Zoom options: All, In, Out, Max, Position.

Curves:
Red: Impulse response timeline

Light blue: 2 segment envelope curve to suppress early reflection and post-reverb.

Yellow: Freely plottable envelope curves

Parameters

Parameter presets
This section lets you save, load, and delete the parameters o the dialog only (response curve, frequency response, mix). This function is most useful when testing different settings without changing the impulse response.

Save/Load/Delete Parameter presets
The load function is only needed if you want to load preset files from different folders than the default folder "FX-preset". The presets for the Parameter that are located in the "FX-preset" sub-folder are automatically loaded and listed in the drop-down menu.
The Parameter presets have the extension *.rsp.

Parameters for editing reverb response times

Plottable envelope curves: The impulse response scale can be manipulated with a yellow freehand envelope curve. Typically, it is used to suppress or delete early reflections. Using the zoom options will help greatly.

Reset Freehand envelope: reinstates the curve's original status.

Pre-delay: Delays the entire impulse response to a selectable time interval from 1..100 ms.

Impulse I.R.

Early Reflect: This parameter allows you to set the amount of the early reflection. It raises or lowers the first part of the impulse response.

Late Reverb: The later reverb portions can be amplified or decreased. This parameter uses the final part of the impulse response.

Length: This setting will shorten the length of the reverb effect to up to 5% of the original length by shortening the impulse response pattern. Please keep in mind that the reverb can end rather abrupt, which may lead to unnatural decay patterns. You can compensate by applying a lower Late Reverb setting to fade the impulse response with the 2 segment envelope curve. The graphic display will assist you in allowing a quick optical check of the envelope.

Env. Time: Sets the duration of the first envelope curve segment.

Parameters for editing reverb frequencies

High: This setting allows dampening of high frequency portions of the reverb effect.

Low: This setting suppresses low frequencies of the reverb effect.

FFT-EQ: The reverb percentage can be re-worked with an additional FFT filter. The real-time spectrum diagram of the reverb percentage in the FFT filter permits optical control of room frequencies by recording the impulse response. Undesired resonance can, for example, be identified and eliminated rapidly. The FFT filter is unavailable with parameter latency settings of less than 4096 samples because it causes greater latency due to its linear phasing function.

Mix

Dry.: Sets the original signal in dB

Wet: Sets the amplitude of the reverb effect in dB

Out: Sets the output level in dB

Bypass: The input is switched directly to the output for comparison.

Performance Options

Quality: In the two normal modes, room simulation is calculated at half the sample rate. This is perfectly sufficient in most cases, since natural or digital impulse responses typically possess components of less than 10 kHz. (you can check this in the integrated FFT filter's spectral editor). Many older reverb devices at half the sample rate – computation above the frequency range that is inefficient use of the CPU.
Both normal modi differ only in the quality of the resample used for sample rate decrease. The "normal" resampling quality is nevertheless quite high. Only in exceptional cases should normal plus mode be used (and in which arithmetic performance increases slightly).

High mode calculates the entire frequency range, doubling the load on the CPU when compared with normal mode.

Retrieve, Set: Saves and set the quality options globally. Different settings are recommended for use in objects, tracks, etc.

Latency: Parameter for block length defaults upon which the folding operation is calculated internally.
Short block length increases the number of arithmetic operations required. This increases the load on the CPU. Large block length leads to irregular CPU load. The optimum for real-time processing tends to be about 32,768 samples. For use in the AUX, track or master, (where lower latency is desired) a setting of 40% is recommended when operating on fast systems.
The quality of the result is independent of this setting.
With latencies of less than or equal to 40%, the FFT filter can no longer be employed. Impulse responses that you have created yourself may apply the FFT filter to the impulse response before destructive room simulation.
The parameter used during room simulation set high by default, since lower latency would increase CPU load unnecessarily.

Set / get Settings: with the get button you can reset these options to its defaults, with the Set button you can define new ones.

Overview: impulse response and parameter tendencies

Quality

"High Quality" setting Double (Factor 2:1)

Latency

Lower latency (2048 samples): approx. increase of 50% (factor 1.5)
Very low latency (128 samples): approx. increase of 150 – 200 % (factor 3 - 4)

Impulse response duration

If the length of the sample used as an impulse response exceeds a power-of-two, the CPU load doubles (factor 2,1).
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.

Make sure that your own impulse responses are shorter than the next value — particularly if the impulse responses are longer than 1 second, because the length of these "continuous CPU load intervals" increases exponentially with each increase in length.

Impulse response sample rates

The CPU load increases proportionally to the sample rate.

Impulse response resolution

The CPU load is independent of it since the computation is carried out with float precision in each case.

Room Simulation: Tips and Tricks

The Room Simulator can be used to recreate the characteristics of a multitude of room reverbs. By editing the impulse response pattern itself a number of different effects can be achieved. Here are a few suggestions:

- Use the Reverse function on the impulse response to create a reverse reverb.
- By using the Timestretching function the size of the room can be altered without affecting the resonance behavior of the room itself.
- Use the Draw function in the Wave window to remove early reflections of the room or add others to it.
- You may want to experiment with short samples of your own audio material by creating a Fade Out in the sample. You can use the sample as the impulse

response. Interesting and sometimes amazing effects can be created this way.
– You can also get interesting reverbs by using the impact on a percussion instrument as an impulse response.

Problems and Remedies

The reverb is too harsh.
The Late Reverb parameter can be used to fade out the impulse response to zero. Use the graphic display to test the parameter's effect on the response. Especially when using the dB log setting for the display, the response can be monitored to see whether the amplitude effectively fades to a zero value.

Resulting file has a large DC offset.
The impulse response must not contain a DC offset. Please remove it by using the menu Effects-> Remove DC Offset.

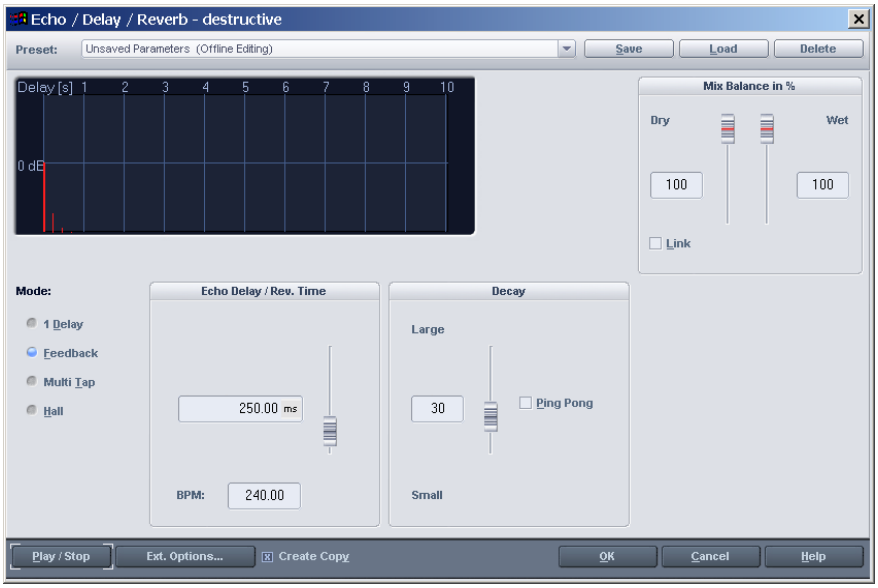
Heavy hard disk activity or an error message concerning insufficient memory was received.
The algorithm needs an immense amount of storage, especially with long impulse response patterns. All background processes should be terminated and all RAM Wave Projects that are not needed should be closed.

The CPU load is too high.
– Check the latency parameter settings according to the advice given in the parameter description.
– Select "Normal quality".
– Reduce the length of the impulse response (while remembering the advice given regarding continuous CPU load intervals).

Problems with very long impulse responses.
When used in real-time, impulse responses of up to 11.88 seconds (with 44,1 kHz sample rate) can be used on up-to-date computers (2002)).
The length is limited only by the CPU capacity.
With destructive computations, the upper limit is 380.4 seconds (with 44,1 kHz sample rate).
If this upper limit is exceeded, the following samples are ignored.
Impulse responses of this length are, however not suitable for room simulation.
If you want to experiment with extremely long impulse responses (longer than 1 minute), you should note that the algorithm will require a lot of main memory (over 500 MB).
When loading long Wave Projects as impulse responses, a warning appears offering the chance to abort the operation to prevent a long wait for loading.

Echo/Delay/Reverb

Choose this option to open the Echo / Delay / Reverb Effect window, allowing you to destructively apply delay effects to Wave Projects.



Note: If this function is used via the Destructive Effects menu, processing is destructive. The same processor is also available for real-time use within the VIP window. To make use of the real-time version, open the Mixer (key M) and right-click on the Delay/Reverb button (on each channel).

Mode

Mode determines the parameters available in the dialog.

1 Delay: This mode sets the processor to function as a single repeat delay.

Feedback: This mode sets the processor to function as a delay that repeats, as it decays to silence. The "echo" events are repeated in accordance with the interval specified in the Echo Delay (Millisec.) settings. Echo decay is applied to the repeated events – as a percentage of the previous event - depending on the settings.

Multitap: This mode sets the processor to function as a multi-tap delay. The multi Tap delay repeats portions of the delay event, between the major delay events determined by the Echo Delay settings. This allows for a variety of delay events, that are repeated and multiply, while the decay "fades" earlier events.

Track Delay: This mode delays the entire Track by a set amount. This can be useful for correcting audio delays in recordings.

Reverb: This mode sets the processor to function as a reverb. Although this mode is not intended to create a world class reverb, the algorithm uses very little CPU power and offers sufficient fidelity for most simple reverb applications.

Echo Delay / Rev. Time

In this field the delay between the individual echoes and/or the original signal and the first delay is displayed in the set measurement unit. The measurement unit is configured by clicking on it and selecting from the menu.
In Reverb mode it controls the Reverb time.

Decay

You can adjust the decay of the delay/multi tap delay effect here. The slider actually controls the amount of feedback from 0 until 100 %, depending from this the delay fades out faster or slower.
Ping-Pong: This applies a Stereo effect to the delay by swapping the channels of the feedback path. To hear this effect, the input signal must

Reverb Properties

Room Size: This fader adjusts the overall size of the reverb chamber. You can also enter the size numerically (with your keyboard) in the adjacent data box.

Dark / Bright: This fader adjusts the overall tonal color (light / dark) of the reverb chamber. You can also enter the tonal color numerically (with your keyboard) in the adjacent data box.

Link: The wet and dry levels are linked to maintain a nearly constant overall signal level when moving one of the faders.

Mix Balance in %

Dry: This fader controls the original signal level.

Wet: This fader controls the level of the Delay's decay (expressed as a percentage of the original audio's level). You can use the Fade to modify the current setting, or type the desired setting directly into the box. Notice that the visual display (lower half of the window) changes to reflect the current level of the Delay's Decay.

Create Copy

As per most of the other Effects menu items, this important option determines whether or not an Undo will be created, prior to processing the file. As always, we strongly recommend that you leave this option enabled.
When this box is checked, Samplitude will create a copy of the selected Range in the audio file that contains the original material. This allows you to compare the results with the original material. Please consider the length of the selected Range, and the available hard disk space or RAM memory.

Load Setup
Click on this button to load a previously saved set of Echo / Delay / Reverb Effect settings.

Save Setup
Click on this button to save the current Echo / Delay / Reverb Effect settings.

Test
This button auditions the processed audio file.

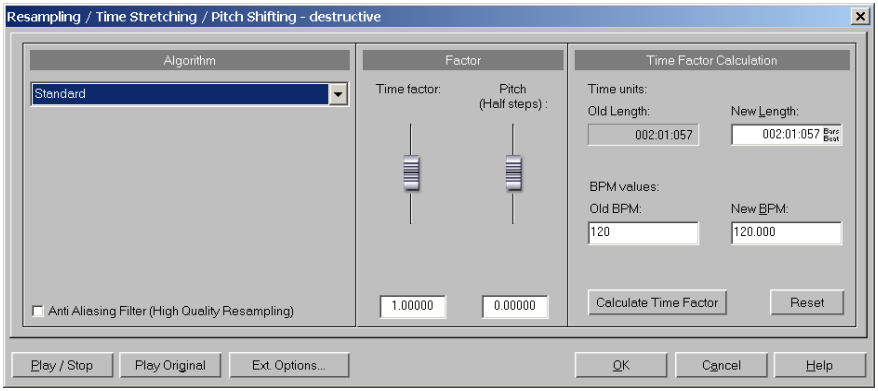
OK
Click on this button to write the Delay/Reverb Effect to the audio file, and close the Echo/Delay/Reverb Effect window.

Cancel
Click on this button to close the Echo / Delay Effect window, without applying any processing.

Resample / Timestretching

Note: If you only want to change the sample rate of an audio file, use the Adjust Sample Rate option from the Effects Menu.

All algorithms in this dialog use the time factor parameter and pitch as input parameter.
The time factor calculation (from the original length/original speed and the length required / speed required) can be comfortably activated on the right side of the dialog.



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. The effect is comparable with changing the playing speed of record players or tape recorders.

The effect is relatively loss-free, the sound loss is smaller than with all Timestretching/Pitchshifting procedures. If a pitch change is justified by changing the speed (and vice versa), use this algorithm.
When increasing the length of the audio material by resampling, very small alias effects may occur. the anti- alias filter helps in this regard. Alternatively, since this filter requires additional CPU load, it can be activated for the computation and the real-time preview by checking "High Quality Resampling". The filter is also available for the other algorithms that use resampling internally when pitchshifting.

Pitch-Shifting/Timestretching

With all other algorithms, pitch and length variation are independently adjustable. These algorithms produce artefacts, and depending upon the audio material, specific algorithms may produce better results. The algorithm used for Timestretching/Pitchshifting as a default Object Effect can be stored for a Wave Project with the timestretch patcher.

Algorithms for Pitchshifting / Timestretching

Standard

An algorithm is used that usually produces very good results. It uses factors (from 0.9 to 1.1) and fixed phases, which maintains the impression of space for stereo signals. For drum loops or other "beat-like" material, this algorithm is only occasionally suitable because it can change the groove and - in rare cases - doubles or erases beats.
Time compression (shorter sample length) usually works better with this as timestretching. When combining two samples, reduce the longer sample rather than the other way around.

Use recommended when:
– working with complex material (Mix)
– the other algorithms fail

Load on the CPU: light

Smoothed

This is a substantially more complex algorithm, which demands more computing time. The material can now be processed even with very large factors (0.2... 50) without creating large artefacts. That's because the material is "smoothed", which creates a softer sound and changes phase position. This smoothing is hardly audible in recordings of speech, singing or solo instruments. With more complex spectra – mixed sounds from different instruments or fixed mixes – problems may arise. This algorithm is not suitable for drum loops and other material with pronounced transience. The groove remains, but the phase shift alters the 'attacks'. With small Corrections (factor approx. 0.9 - 1.1), a smallest possible Smooth degree often does the trick. You can adjust the smooth degree in five steps.

Use recommended for:
– orchestral instruments
– speech recordings with background noise (e.g. video recordings)
– Synth surfaces, guitars...

Use not suitable for:
– Stereo mix
– Drum loops, percussion

Load on the CPU: very heavy

Beat marker-based slicing
This mode was primarily conceived for adjusting drum loops, but can also be used for other material such as monophonic bass lines or sequencer lines. The algorithm divides the material into individual components – individual notes or beats marked by beat markers.
The beat is then reconstructed using this "snippets". Individual beats are overlaid when the tempo is increased. Small pauses appear between beats when the tempo is decreased.
If you are able to use this algorithm, do so. It allows you to change the tempo without loss of quality and with minimum CPU load.

Use recommended for:
– Samples to be cut into individual notes or bars (beats)
– Drum loops, but only when beats don't overlap and there is little reverb

Use certainly not suitable for:
– Anything else

Load on the CPU: very light

Beat marker-based stretching
This algorithm works like the standard algorithm, but synchronizes the stretched material with the beat markers. The groove is consequently altered somewhat, and beats may be duplicated or deleted. In contrast to the beat marker-based slicing algorithm (which may produce missing or incorrectly positioned markers), this algorithm's disadvantage lies in decreased synchronization performance. In practice, the automatic generation of markers almost always produces good results. One should avoid placing beat markers too close to each (< 1000 samples.)
„Long overlap": With this option active, the sounds are less "roughened up", but echoes can appear instead. This is especially suitable for drum loops with very little time changes.

Use recommended for:
– All types of rhythmic samples that can't be cut into individual notes or measures (beats)

– Complex overlapping drum loops
– For when beat marker-based slicing can't be employed or when it's too much bother to find the perfect beat marker position.

Load on the CPU: light

Beat marker-based slicing and beat marker stretching – Setting and evaluating beat markers

If there are no "patched" (see below) beat markers inside the Wave Project, these are determined automatically in real time. If there are beat markers, you can determine by the option "patched" if they are used or not.
In the offline-effect dialog for timestretching you can temporarily create such markers for offline processing. To save beat markers permanently into the Wave Project and for more comfortably editing, use the Timestretching patcher (menu Tools)

Generating beat markers — Options:
Find beat marker: The algorithm searches for and marks out beats in the audio material.

Beat recognition threshold: Sets the threshold for beat recognition. When the Wave Project is being displayed, you can follow the placement of beats. This also works with Objects if the Wave Project is kept open simultaneously.

Reset Beat markers: All beat markers are removed.

Beat markers can be re-positioned, copied, deleted etc. by activating the applicable marker commands.
When new markers are created, the marker name must be assigned "Beat Marker". Before editing, we advise that you set the beat recognition threshold to a high value and then delete the redundant markers. It is quicker than adding markers manually.
You can only edit markers while the dialog is open by using the Timestretching-Patcher. There you can edit the beat markers without closing the timestretching dialog (non-modal window).This allows you to check the results without having to close the dialog.
Furthermore you can save the tempo (BPM) and the preferred method for Object-based timestretching directly to the Wave Project.

Monophonic Voice

This is Timestretching and Pitchshifting for solo vocals, speech or solo instruments. The material must not contain background noises and should be free of reverb. With suitable source material, the audio quality is very high. The "Use formant correction" option secures the formants when pitchshifting so that the "Mickey Mouse effect" doesn't occur. You can therefore "compose" a background chorus from just one solo voice. Formants can also be shifted across + - 12 half tones to create voice distortion.

Typical uses are:
- Intonation correction (imprecisely pitched notes should be removed as Objects and worked on separately if not working with ElasticAudio)
- Harmonizer effects: an Object with singing can be copied and shifted downwards. If the pitch is then changed, a second voice is generated, etc.
- Creating background vocals from existing vocal samples
- Timestretching
- Speech sample distortion

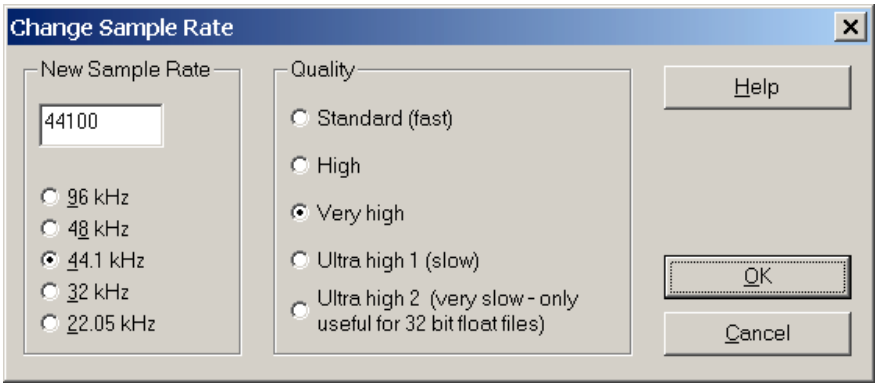
Use recommended for:
solo vocals, speech or solo instruments without overlapping, little or no reverb or background noises.

Use certainly not suitable for:
Anything else

Load on the CPU: heavy

Change Sample Rate

This option opens the Change Sample Rate window, allowing you to convert the sample rate of Wave Project. As an example, this is where you convert audio from 48kHZ to 44.1kHz, in preparation for CD burning. This process will create a new copy of the Wave Project file, and apply the Sample Rate Conversion to the copy. In other words, you don't have to worry about destroying the original audio file.



Important note: You can only convert the sample rate(s) of Wave Project files! If you wish to convert the sample rate of an Object in the VIP window, right-click on the Object and select Wave Editing. This will open the Wave Project file that is referred to by the Object.

New Sample Rate

This parameter determines the new sample rate for the copy (Wave Project). You can use the radio buttons to select from the most common sample rates, or you can type the sample rate into the box. *Your audio card MUST support the new sample rate, or you will not be able to play the converted Wave Project!*

Quality

This parameter determines the degree of Anti-Alias filters and interpolation used when converting sample rates. The higher the quality, the longer the sample rate conversion takes to process.
The available conversion algorithms range from Standard to Ultra High 2, offering a suitable quality for all needs:

Standard

An interpolation algorithm is used, which is also applied during realtime resampling. This setting is appropriate for work where high quality is not necessary, but speed is important.

High

This adds a filter 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 kHz signal, this setting may improve the results.

Very High

This setting uses a new algorithm, which utilizes a higher sample rate internally. The input and output sample rates are divisible by this internal sample rate. (As an example, the conversion from 48 to 44.1 kHz uses an internal sample rate off 7.056 MHz). This ensures the highest possible quality. This method is also used during CD-Bouncing, if the sample rate of the Project is not 44.1 kHz.

Ultra High 1

Similar to Very High, but with a higher internal filter order. CPU load increases dramatically.

Ultra High 2

Similar to Very High, but with a super high-quality internal filter order. CPU load increases sharply. This algorithm is only useful for non 16Bit files, as the maximum level possible for artefacts is lower than -96 dB.

Note: Reducing the sample rate will result in a loss of high frequency content, i.e When converting from 48kHz to 44.1kHz, this loss is negligible and above the range of human hearing. When converting from 44.1kHz to 22.05kHz, however, you will definitely notice the difference. The highest frequency that can be represented by the audio is equal to half the sample rate. In other words, if the sample rate is 22.05kHz, the highest frequency that the audio can reproduce is 11.025kHz.

OK
Click on this button to convert the sample rate of the audio file (using the current settings), and close the Change Sample Rate window

Cancel
Click on this button to close the Change Sample Rate window, without applying sample rate conversion.

Some Notes On Sample Rate Conversion (SRC):
Sample Rate Conversion (SRC) is often misunderstood by newcomers to digital audio. Some people become confused about the difference between SRC and Bit Depth (16/24/32 Bit). Here are a few quick notes which broadly summarize the important issues:

1. Sample Rates are principally related (by the Nyquist Theory) to frequency response, while bit depth is principally related to Dynamic Range. In other words, sample rates have no effect on Dynamic Range.
2. Anti Alias Filters are used when performing SRC to reduce Alias Distortion, while dithering is used when reducing bit depth to reduce (mask) quantization distortion, caused by truncation (removing bits from the word length i.e. 24 bit truncated to 16 bit).
3. A simple chart: sample rate > Frequency Response > Anti Alias. *Resolution=Bit Depth > Dynamic Range > Quantization > Dithering.*
4. Due to the complicated mathematics involved in SRC, the quality of the conversion was - at best - a compromise in the past. Poor interpolation algorithms, combined with the steep filters required for the removal of alias distortion, often resulted in noticeable quality differences between the original and the resampled audio. In recent years, advances in algorithm quality have dramatically reduced such artifacts. The algorithms in Samplitude (particularly the Ultra settings) offer extremely high quality SRC results. We are sure that even the most pedantic of audio engineers will be very impressed with Samplitude's SRC capabilities!
5. In the past, some audio engineers would perform SRC by re-recording the audio, in the analog domain at the new sample rate, to avoid the problems of poor SRC. As an example, a 48kHz mix needs to be converted to 44.1 kHz for CD Mastering. The engineer would play the mix at 48kHz through the analog converters, while re-recording the analog audio at 44.1 kHz onto another recording device.

By using this method, digital SRC is avoided and therefore Alias distortion is avoided (although the filters in AD/DA converters also create some distortion around the Nyquist frequency).
This is still a legitimate method of performing SRC, particularly if your AD/DA converters are of the highest standard. If you're one of the engineers who employ this method, we urge you to try Samplitude's high quality settings, and carefully compare results. You will be pleasantly surprised at the excellent

SRC Samplitude offers – and will save a LOT of time when converting your projects.

Sample Manipulation
Sample Manipulation → Sample Data /2

Every second sample is removed, and the entire sample is reduced to half its length. The audible pitch is doubled, i.e. raised by one octave. When halving the sampling rate, you will notice that the corresponding upper harmonics are missing. This option is only available when working in Destructive Editing mode - i.e. directly on an HD Wave file.
The frequency Range is divided (i.e. 0-22.05 kHz @ 44.1 kHz sample rate – > 0-11.025 kHz @ 22.05 kHz).

Sample Manipulation → Sample Data *2

This function inserts a new value between two neighboring sample values. It is the average of the two sampling values.
The entire sample is doubled in length, and the resulting pitch of the sample is halved. You must then double the playback rate to achieve the former pitch. You should note that the higher sample rate does not lead to new upper harmonics – the frequency range of the audio material remains the same.
This option is only available when working in Destructive Editing mode - i.e. directly on an HD Wave file.

Sample Manipulation → Reverse

The sample data in the selected Range is reversed along the time axis - i.e. the file will play backwards, with the (former) end position now placed at the (former) start position, and vice-versa. This allows for very interesting effects, not to mention 'backward masking' hidden messages, frequently rumored to be in various songs....
This function is reversible: if you do not select a new Range, reuse of the function restores the original material.
The option is only available when working in Destructive Editing mode - i.e. directly on an HD Wave file.

Build Physical Loop

This function features a complex loop optimization algorithm, for use in Wave Projects. It is useful when samples are to be used for instrumental sounds, or for Wavetable synthesis. Sample loops which feature strings, drones, pads and ambient effects (all usually difficult to transparently loop) can easily be created with this option!
Prior to processing a sample, you will need to select a Range that defines the approximate boundaries (edges) of the sample loop. To do this, you must be in Destructive Editing mode - i.e. working directly on the HD Wave Project. (*Object > Destructive Editing* when working in a VIP)

Remember that you can shift and vary a Range during playback, in order to find the best loop position. A good shortcut technique for fine-tuning loops positions is provided

by activating the split Range mode (Shortcut – B). The sample will now be displayed in 3 sections.
To ensure a smooth, glitch-free loop, the outer limits of the Range will be set to zero amplitude. By applying a crossfade to the material at the loop endpoint (containing the sample data in front of the loop's start point), Samplitude will create a transparent transition between the loop end and start points.
When a marker is placed before the selected Range, the Range between the marker and the loop start point will be used for the crossfade. This offers you a feature found in professional hardware samplers. To achieve a short crossfade, set the marker closer to the loop start point. To achieve a longer crossfade, position the marker further away from the loop start point.

Note: The distance between the marker and start point of the loop Range needs to be smaller than the loop Range itself, in order to make a crossfade possible.

Direct X / VST Plug ins

This function allows you to use Microsoft DirectX and VST 2.2 compatible plug-ins with Samplitude. This complements the already existing superb effects with an unlimited number of 3rd-party effects.
The Mixer and the Object Editors in Samplitude are also compatible with plug-ins, which can be used as real-time effects. However, the plug-ins have to be able to process the audio data blocks immediately and completely, without changing the length of the material.
This is the case for most plug-ins. Other algorithms will fail, such as time stretching plug-ins, since they are based on changing the length of the material. The menu Effects does not pose these limitations. All available plug-ins should work, even if they change the length of the audio material or temporarily off-load the data blocks.

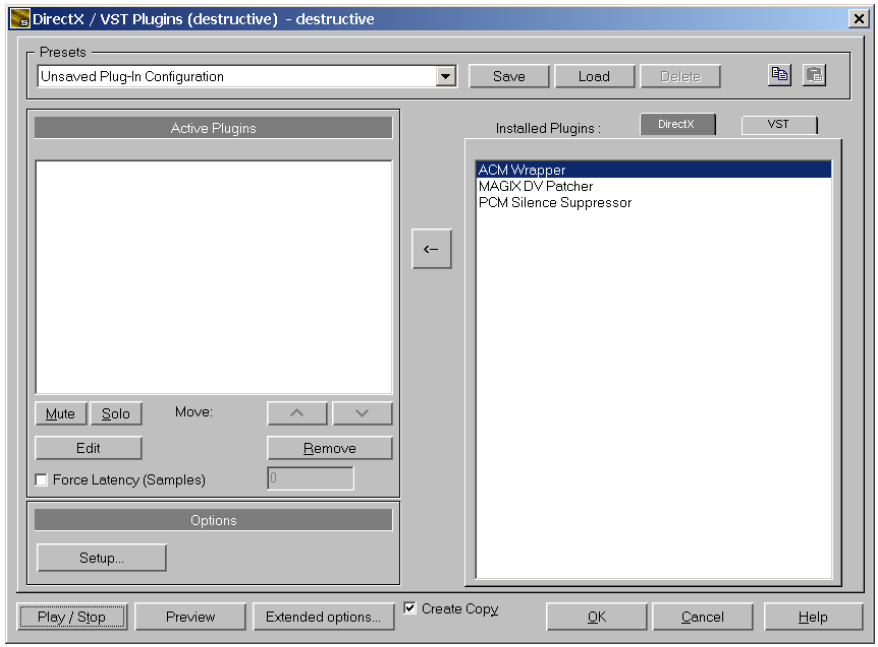
Working with the Plug-Ins

After opening the plug-in dialog, a list of installed plug-ins is visible. Double clicking on a specific plug-in moves the plug-in to the left side of the dialog. The left side shows the active plug-ins. At the same time the plug-in is moved into the active plug-in list, the dialog for the chosen plug-in is displayed. The Plug-in dialog allows you to make further settings for the effect. Additional double clicks on entries in the right-side list add other plug-ins to the active plug-in list on the left side. Please make sure that the chosen plug-ins are compatible with each other. For example, mono and stereo plug-ins cannot be used simultaneously – an error message is displayed. In this case you can "force mono processing".

Pre/Post section (Mixer only)
In the Mixer, the list of the active plug-ins is divided in two sections, pre and post. In the FX routing page of the track or the master section there are four corresponding entries, DirectX plug-in Pre and Post and VST Plug-in pre and post. So you can use a plug-in (or more as one) at two different places of the

effects chain. All plug-ins loaded below the pre line in the list are inserted in the effects chain at the pre position, all below the post line at the postposition.

Note: By technical reason it is not possible to put DirectX Plug-ins after VST Plug-ins, if they are used together in pre or in postposition. To achieve this, put the VST Plug-in pre and the DirectX Plug-in post.



Dialog Elements

Play/Stop: This button activates the real-time preview of the active plug-in listed on the left side of the display. This function is ideal for testing of the chosen plug-in settings if the real-time calculation operates sufficiently.

Preview (only in Effects menu): This function calculates a short segment of the audio material with the active plug-in settings and plays back the audio segment.. Use this option if your system does not seem to be able to sustain the real-time preview ("Play/Stop' button). The length of the off-line preview can be determined with the setting System (shortcut: y) – Test buffers.

Create Copy (only in Effects menu): This decides whether Samplitude creates a copy of the original material. If the option is checked, the copy is appended to the end of the audio file and the effect is processed over the copy instead of the original. This function allows the Undo option to return the Object to the original state. It is therefore recommended to always use this option!

Solo: Only the selected Plug-In will be used.

Mute: The selected Plug-In will be muted (bypass function). You can also bypass a plug-in by unchecking the preceding checkbox.

Remove: the selected plug-in will be removed from the effects chain.

Arrows: moves the plug-in up / down in the effects chain

Plug-in latency: Shows the reported latency of a Plug-in. With Force Latency you can override this value, in case a wrong latency was reported from the Plug-in.

Setup
Write DirectX log file: For trouble shooting problems with a special plug-in you can activate logging here.

Start all Object related Plug-Ins at playback start: This option is only needed when using Plug-Ins with the Object Editor. Some Plug-Ins causes playback errors if they are started independently. The errors occur when the Object with the assigned Plug-In is played back. You can prevent this by choosing this option. At the other hand, the performance will significantly slow down, because Samplitude has to calculate all plug-ins, no matter if the Object which uses it is actually playing.

Disable DirectX while Scrubbing/Jogging: This is checked by default, because some popular plug-ins cause problems while scrubbing.

Presets
Saving and Loading: These two buttons in the upper left-hand section of the dialog allow you to save and load Plug-In presets. A previously saved preset, which is saved in the standard FX-preset folder will be listed in the selection box. A Plug-In Setup contains the information of utilized plug-ins, all the parameters, and the order in which the plug-ins are set.
This allows you to easily transfer custom settings between several instances and projects in Samplitude (Mixer, Object Editor, Offline Wave editing in the Effects menu).

Copy/paste: With the copy/paste button in the top right corner you can even quicker transfer plug-in settings between different instances of a plug-in chain.

Some Tips for Working With Plug-ins in the Effects Menu

1. When using plug-ins which require mono file input (e.g. Antares Autotune) on a stereo file, you must convert the file to mono before using the plug-in. (*File > Save In Format*) After converting to either a single, or two mono tracks (a piano recorded in stereo, for example), you can then open the mono file/s, process it/them with the plug-in, and re-import the processed file into the project (usually by click-dragging the selected file into a track).

Note: This is slightly different when using DirectX in the VIP (i.e. in realtime). In this scenario, the Force Mono Processing box must also be checked.

2. It is unnecessary to leave plug-in windows open, once you’re happy with a setting. As an example, if applying multiple plug-in effects to a file, you could end up with plug-in windows all over the screen. To avoid this situation, simply close each DirectX plug-in dialog once you’ve made your settings. Samplitude will retain the settings. You can reopen these dialogs at any time, for further adjustment, by double-clicking on the appropriate entry in “the left-hand column of Samplitude’s plug-in dialog. Once all settings are to your taste, click the OK button.

Process only left/right stereo channel

Only the right/left channel of a stereo Wave Project will be affected by effects, when ”Left/Right processing only” is used. This option is only available when in Wave/Destructive Editing mode.
The L/R buttons (in the lower left corner of the Wave Project window) have the same effect.

"Range" Menu

The Range menu contains functions which enable you to select sections (Ranges) of a project in various ways. Ranges work in VIP tracks, and when working directly on Wave Projects.
Many of Samplitude's processing options require you to select a Range prior to their use. Spending some time learning the many ways you can select Ranges in Samplitude, will dramatically increase your editing speed.

Note: When selecting a Range for playback (when looping a particular section for overdub rehearsal, for example), it is unnecessary to drag the Range vertically down over all of the tracks. Selecting one track will still result in the playback of all tracks.

Range all

Choose this option to quickly mark a Range over the entire Project (Wave project or VIP).

Shortcut: *A*

Move play cursor → to Beginning

Choose this option to position the play cursor at the beginning of the project (Wave Project or VIP).

Shortcut: *Home*

Move play cursor → to end

Choose this option to position the play cursor at the end of the project (Wave project or VIP).

Shortcut: *End*

Move play cursor → to Range start

Choose this option to position the play cursor at the beginning of the currently marked Range.

Move play cursor → to Range end

Choose this option to position the play cursor at the end of the currently marked Range.

Move play cursor → to Section start

Sets the start position of the play cursor to the beginning of the screen section.

Move play cursor → Left/Right Move in Page/Scroll mode

With this command you move the play cursor left and right. In Page mode the project window is scrolled page-wise. When the play cursor arrives at the project window border, the section is shifted, the play cursor runs into the new section

from the opposite side. In Scroll mode the Playcursor stays constant, the project is moved under the playcursor.
If there is a Range selected, moving the play cursor in Page mode moves also the Range start.

Shortcut:	*Left Move in Page mode*	*Left arrow*
	Left Move in Scroll mode	*Alt + Left arrow*
	Right Move in Page mode	*Right arrow*
	Right Move in Scroll mode	*Alt + Right arrow*

Note: The above functions work best when using the keyboard shortcuts.

Move play cursor → Object border left/right

This option moves the play cursor to the left (right) border of the selected Object.

Shortcut:	*Object border left*	*Ctrl + Q*
	Object border right	*Ctrl + W*

Move play cursor → Marker left

This option moves the play cursor to the next marker to the left (previous marker).

Shortcut: *F2, Alt + Q*

Move play cursor → Marker right

This option moves the play cursor to the next marker to the right (next marker).

Shortcut: *F3, Alt + W*

Edit Range → Move Range start left

Moves the start of the Range one pixel to the left in the current VIP window. The distance moved is determined by the zoom factor. The greater the zoom factor, the shorter the distance.

If the playback is not running, the command is the same as Move play cursor Left/Right. Use this command especially while playback is running to move the Range start without affecting the current play cursor position!

Shortcut: *Alt + "/" (Num pad)*

Edit Range → Move Range start right

Moves the start of the Range one pixel to the right in the current VIP window. The distance moved is determined by the zoom factor. The greater the zoom factor, the shorter the distance.

If the playback is not running, the command is the same as Move play cursor Left/ Right. Use this command especially while playback is running to move the Range start without affecting the actual play cursor position!

Shortcut: *Alt + "*" (Num pad)*

Edit Range → Move Range end left

Moves the end of the Range one pixel to the left in the current VIP window. The distance moved is determined by the zoom factor. The greater the zoom factor, the shorter the distance.

Shortcut: *Alt + "-" (Num pad); Shift + <- (Cursor left)*

Edit Range → Move Range end right

Moves the end of the Range one pixel to the right in the current VIP window. The distance moved is determined by the zoom factor. The greater the zoom factor, the shorter the distance.

Shortcut: *Alt + "+" (Num pad); Shift + -> (Cursor right)*

Edit Range → Range to beginning

Choose this option to extend the currently marked Range to the beginning of the project (Wave Project or VIP).

Edit Range → Range to end

Choose this option to extend the currently marked Range to the end of the project (Wave Project or VIP).

Edit Range → Flip Range Left

Choose this option to flip the currently marked Range to the left. (What used to be the start of the marked Range is now the end of the marked Range.)

Shortcut: *Ctrl + Shift + Left arrow*

Edit Range → 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*

Edit Range → Beginning of Range → 0

Choose this option to move the beginning of the currently marked Range to the next zero amplitude crossing.

Tip: Editing Cuts should be made at zero amplitude crossings, to prevent clicks. Samplitude's auto crossfade features will usually prevent this problem from being

an issue when editing. If you are cutting audio for use in another software application, however, ensure that you place your cuts at zero amplitude crossings.

Shortcut: *Ctrl+Page Up*

Edit Range → End of Range → 0

Choose this option to move the end of the currently marked Range to the next zero amplitude crossing.
"Zero" is the next sample value with zero value or the boundary between a positive and a negative sample value (or vice versa).
In VIPs this works too, but Objects have to be present somewhere in the VIP in the selected Range.

Tip: Editing Cuts should be made at zero amplitude crossings, to prevent clicks. Samplitude's Auto Crossfade features will usually prevent this problem from being an issue when editing. If you are cutting audio for use in another software application, however, ensure that you place your cuts at zero amplitude crossings.

Shortcut: *Ctrl+Page Down*

Edit Range → Beginning of Range ← 0

Choose this option to move the beginning of the currently marked Range to the previous zero amplitude crossing.
"Zero" is the next sample value with zero value or the boundary between a positive and a negative sample value (or vice versa).
In VIPs this works too, but Objects have to be present somewhere in the VIP in the selected Range.

Tip: Editing or Cuts should be made at zero amplitude crossings, to prevent clicks. Samplitude's Auto Crossfade features will usually prevent this problem from being an issue when editing. If you are cutting audio for use in another software application, however, ensure that you place your cuts at zero amplitude crossings.

Shortcut: *Shift + Page Up*

Edit Range → End of Range ← 0

Choose this option to move the end of the currently marked Range to the previous zero amplitude crossing.
"Zero" is the next sample value with zero value or the boundary between a positive and a negative sample value (or vice versa).
In VIPs this works too, but Objects have to be present somewhere in the VIP in the selected Range.

Tip: Editing or Cuts should be made at zero amplitude crossings, to prevent clicks. Samplitude's Auto Crossfade features will usually prevent this problem from being an issue when editing. If you are cutting audio for use in another software application, however, ensure that you place your cuts at zero amplitude crossings.

Shortcut: *Shift + Page Down*

Edit Range → 0 → Range ← 0

Choose this option to move the beginning of the currently marked Range to the next zero amplitude crossing, and to move the end of the currently marked Range to the previous (left) zero amplitude crossing.

Shortcut: *Clear*

Note: If you do not have a Clear key on your keyboard, it is easy to assign your own shortcut, for this option. A possible suggestion is Ctrl+Shift+Up arrow, as this is not used by default. To change the shortcut key, open the Options > Program Preferences > Edit Keyboard commands window.

Edit Range → Range start to left marker

Choose this option to extend the Range to the next marker on the left.

Shortcut: Shift + F2

Edit Range → Range start to left Object border

Choose this option to extend the Range to the left border of the next Object.

Edit Range → Range end to right marker

Choose this option to extend the Range to the next right marker.

Shortcut: Shift + F3

Edit Range → Range end to right Object border

Choose this option to extend the Range to the right border of the next Object.

Edit Range → Range over all selected Objects

Choose this option to extend the Range over all selected Objects.

Range length to

The following options allow you to adjust the length of a marked Range to a precise number of beats.

Note: To define the length of a bar; select View > snap and grid setup.... You can now enter the time signature, tempo (in beats per minute), and pulse per quarter note resolution.

1- 16 Beat(s): Choose this option to adjust the length of the currently marked Range to a duration of 1, 2, 4, 8 or 16 beats.

Tip: The grid option (under the View menu) can display bars and beats, and the snap function can be set to "snap to bars".

Split Range

Choose this option to display the current project (Wave Project or VIP) in three independent views. The top view shows the entire project (Wave Project or VIP), the lower left view shows the audio near the beginning of the currently marked Range, and the lower right view shows the audio near the end of the currently marked Range. Each view has its own independent zoom controls, and can be used to manipulate the currently marked Range.

Note: To return the screen to a normal view, press Shift + B.
Tip: This option is very useful when trying to create loops! In the lower views, you can zoom-in (independently) at the beginning and end of the currently marked Range. This allows very detailed adjustments to the Range borders, without having to constantly zoom in and out.

Shortcut: B

Split Range for Video

This option is nearly identical to the Split Range option above, but is intended for use with AVI (video) files.

Note: To return the screen to a normal view, press Shift + B.

Store Range

In Samplitude you can define and store different Ranges for future retrieval. An unlimited number of Ranges can be defined. You can for example specify different loops and compare them while you recall their Ranges.
All Ranges of a project can be seen in menu Tools-> Manager->Range Manager. There all Ranges can be renamed or played.
When selecting this menu option, the selected Range needs to be stored by entering a numerical value the Range is associated with.

Shortcuts:
Keys: ALT + F2 ... F10

Other....

You can define more Ranges by using the submenu "other". You will need to specify a name for the Range selected.

Shortcut: Alt + F11

Note: Ranges stored under specific names can only be recalled with the Range Manager (Tools menu).

Get Range

By selecting this option, you can choose one of the stored Ranges as the current Range.

Samplitude even lets you set a Range while playing. The specified Range becomes the current one and is audible. Using this method, you can change between two Ranges comparing them with each other.
Ranges can also be named and recalled with the Range Manager (Tools-> Managers menu).

Shortcuts: *Ctrl+F2 ... F10*

Get Range length

The following options are similar to the Get Range options, but in this case only the stored Range's length is recalled (not the Range itself). This Range length is then used to mark a new Range starting from the current play cursor position.

Shortcuts: *Ctrl + Shift + F2....F10*

Store marker

The following options allow markers to be placed, and their location stored (for later recall and positioning of the cursor).

Other....

Choose this option to place a marker (and store its position) under a specific name.

Note: The position of markers stored under specific names can only be recalled with the marker/CD Track Manager (Tools menu).

Shortcut: *Key "?"*

Auto Number

Choose this option to place an automatically named marker (and store its position).

Shortcut *Key "'"*

Marker on Record Position

With this option, which is enabled by default, you can set markers while recording at the current recording position. You can disable this behavior here, then also while recording the markers are set at the current play cursor position. Keep in mind, that Playback and recording are independent functions, that means you can play back the beginning of a just recorded Object while still recording it!

1 – 10

Choose this option to place a marker (and store its position for later recall) to the 1....10 key.

Shortcuts: *Shift + 1....0*

Note: all markers can be placed in real-time during playback!

Defined markers can be seen above the sample data section of the project window in the marker bar and can be moved with the mouse. Right-click on a marker to open a context menu with the most important menu entries concerning markers. Double click between two markers on the marker bar marks a Range between the markers.

Get marker

The following options allow marker positions to be recalled in real-time. You can also recall marker positions using the marker/CD Track Manager or the Audio marker Manager (Tools menu).

1 – 10

Choose this option to recall the marker position stored to the 1...0 key. The cursor will be placed at that position.

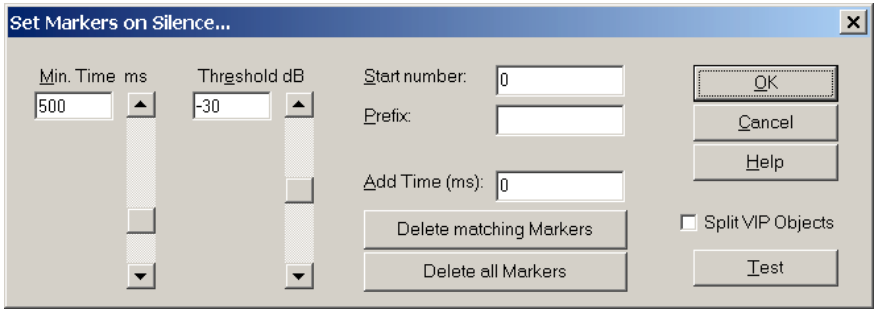
Shortcuts: *1....0*

Marker on Range Borders

Choose this option to place markers at the start and end points of the currently marked Range. The markers will be labeled S (start) and E (end).

Set markers on silence....

This option opens the Set Markers on Silence.... window. This allows you to have Samplitude automatically detect silence, and create/place a marker each time silence is encountered. The "silence threshold" can be defined. When working in a VIP, you can set Samplitude to split selected Objects wherever silence is encountered.



Important Tip: Only audio to the right of the play cursor, or audio within a marked Range will be searched! If you wish to search the entire audio file, press the HOME key to place the play cursor at the beginning of the file.

Min. Time (ms): This parameter determines the minimum length of silence that must be encountered before the algorithm will actually consider it "Silent", and create/place a marker.

Threshold (dB): This parameter determines the level at which the audio is considered silence.

Tip: To quickly see all locations that are below a certain level, use this parameter in conjunction with the Test button (see below).

Start Number: This parameter defines the starting number used when placing the markers. As an example, if this parameter is set to 5, the first marker will be labeled 5, the next marker labeled 6, etc.

Prefix: This parameter allows you to enter a prefix that will appear before each marker number. As an example, if you enter "low" for this parameter (and 5 for the start Number parameter), the first marker placed will be labeled low5, the next marker labeled low6, etc.

Add Time: After finding silence the marker will be moved through, adding a positive or negative time in ms.

Delete Matching markers: Click on this button to delete all markers that meet the criteria set by the min. Time and threshold parameters. As an example, let's say markers were placed by a previous search. If you enter the same criteria (set by the min. time and threshold parameters) and click on this button, these markers will be deleted.

Delete all markers: Click on this button to delete all markers in the window, or currently marked Range.

OK: Click on this button to perform the search, place the markers, and close the dialog.

Cancel: Click on this button to close the dialog without performing a search and creating/placing markers.

Split VIP Objects: When working in a VIP, select this option to have Samplitude split the selected Object(s) wherever a silence marker is placed.

Test: Click this button to preview the results of the function, prior to completing the command. This is useful for checking that the threshold settings are suitable for your needs. Once you are happy that the threshold settings are right, click the OK button to commit the changes.

Note: Split Objects are not shown until you commit to the changes.

Marker Tips: You can click and drag a marker to move it. You can also set and adjust various markers via the markers button, at the top of the Track Properties section (see screen shot below), or by right-clicking on a marker.

Comparisonics Audio Search

This function uses the patented Comparisonics algorithm. This allows you to find the same, or similar sounding, regions in an audio file very quickly – a revolutionary feature!
The function has a variety of uses including:
- quickly finding particular, repeated, sounds in a recording.
- forensic audio work.
- finding repeated animal (or other) noises on lengthy field recordings.
- finding instances of a particular instrument in a classical recording (e.g. a cymbal)
- finding examples of repeated Foley, or other distinctive sounds, when editing pre-mixed audio for film/video.

It is assumed that a HD Wave Project is already open. If not, select an Object in the track you wish to search, then select *Object > Wave Editing*.
Prior to selecting the command via the menu, you must first select a Range (to be used as the basis for finding other "similar" Ranges), and copy it into the clipboard (Key "C"). The Range can be of any duration, but 5ms – 5 secs will provide the best results, depending on the dominant frequency of the audio you are searching for.

Note: The waveform display does NOT need to be set to Comparisonics Colors (View > VIP Display mode > Definitions) for this function to work. In fact, with the Comparisonics Colors displayed. you would rarely need to use this function, as similar areas can be identified by color!

Once you have selected the Range and copied it to the clipboard, launch the dialog from the Range > *Comparisonics Audio Search* menu.
The dialog allows you to change the sensitivity for the algorithm. During the adjustment of the sensitivity level, Samplitude displays "Match markers" at positions identified as a match for the material in the clipboard.
In the example above, the algorithm has found 4 locations which contain audio similar to that of the clip (which, in this case, contained a wide frequency spectrum including bass guitar, kick drum and crash cymbal). note that the original Range area has been correctly identified by the algorithm. The sensitivity fader is used to adjust the threshold of the algorithm, thus determining the relative similarity to the audio in the clipboard. You may need to experiment a little with this setting to determine the appropriate position.
In the example shown above, the algorithm correctly identified the complex frequency spectrum of the clipboard audio, and the four times the particular section of audio appeared in the file (a four bar sample of a rhythm section). Other areas of the file were not significantly different (the file is four bars of bass guitar and drums), yet the algorithm managed to select the correct sections.

Note: The audio copied to the clipboard can also be used as the sample for finding similar sections of audio in other Wave Projects. Simply open the other file(s) you wish to search, and use the function.

Copying audio marker to VIP marker

Creates copies of all available audio markers of the selected Object in the VIP. The Audio Markers of the relevant Wave Project are adopted only within the Object boundaries.

Copying VIP marker to audio marker

Copies the markers in the VIP to the corresponding position of the Object within the Wave Project. Adoption of the VIP Markers occurs within the Object boundaries only.

Delete marker

This function deletes the currently selected marker. To select a marker, either; use one of the "Get marker" commands, or position the mouse cursor over the marker until the bi-directional arrow cursor appears. Once visible, click on the marker.

Note: When using the mouse to select the marker, the quickest deletion method is to select the marker as shown above, then press delete on the keyboard. You can also right-click on the selected marker and use the delete marker command.

Delete all markers

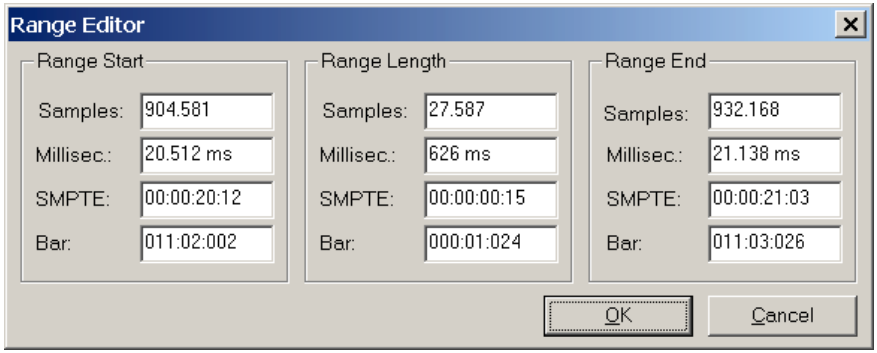
This function deletes all markers in the active project. The command does not simultaneously delete markers from the VIP and all Wave Projects referenced by the VIP Object(s) – the command must be applied separately to each Wave Project. In other words, deleting markers from the VIP does not delete markers from Wave Projects.

Recall last Range

Choose this option to recall the last Range that was marked in a Wave Project or VIP window. As an example, if you've mistakenly removed a marked Range, selecting this option will restore it.

Shortcuts: *Shift + Backspace*

Range Editor



Choose this option to open the Range Editor window.
This window allows you to specify the currently marked Range's start and end positions, and length (in Samples, Milliseconds, SMPTE, or Bar formats).
If you change any of the values in the "Range start", "Range length" or "Range end" sections, all other values will automatically be updated, with the following exceptions:
Change the Range start value – the end point will be retained.
Change the Range end value – the start point will be retained.
Change the Range length value – the start point will be retained.
Please note the format of the "Bar" dialog box. The format is displayed in 4/4 time, with a resolution of 96 clicks per quarter note. Bar, beat, and clicks are displayed.
The number of beats per minute (BPM) can be set in the BPM dialog box, accessible via the View > Grid Setup option.

Range Manager

Please refer to the chapter "The Managers"

Object Lasso

The normal way to select several Objects at once would be to click on the free space on the right of the Object in the lower area of the track and to pull a frame around the selected Objects.
Sometimes, however, there is not enough space between the Objects. The Object lasso solves this problem.
Once activated, you can click on an Object and pull up a selection frame, without moving the Object you clicked on (which would be the case in the standard mode). Once you release the mouse left-click, the mouse returns to the selected mouse mode.

Edit Time Display

Shows or hides the time display window.
This window always shows the current time position in the current format. You can change this format with the menu "Units of Measurement". We

recommend the SMPTE format: hours:minutes:seconds:frames. The colors and the font for the display can be changed in the File->Preferences. You can zoom the window to any size and position it anywhere on the screen.
Click the right mouse button on the Time Display to display a pop-up menu that lets you choose the preferences for the Time Display.
This popup menu contains several shortcuts for program options to customize the font as well as the colors for foreground and background of the display. You can also select the time unit here, it is the same unit as defined in the View menu.
You can set up from one to five rows with "Number of Rows". Each of the rows can display one of the following values, which you can select in the "Current field menu" by right-clicking in the according row.

St. - Position/Range start
This line shows the current Range start position or the current real-time cursor position.

Len - Range length
If a Range is selected, this line displays the length of the Range.

End - Range end Position
If a Range is selected, this line displays the Range end position.

CD Positions
This shows several CD related time positions, including CDS: position from CD start (first set Track Marker), CDE: position to CD end (distance to CD End Marker), CDT: position inside the actual CD track or TrE: position from actual CD track end.

Act - Current Mouse Position
This line shows the current mouse position (not editable).

Mix – Latest Mixer Position
This line shows the value of the recently selected mixer control (not editable).

Rec- Current recording position
This line shows the current recording position (not editable).

RSp- Remaining disc space for recording
This line shows the remaining recording space.

In the same way you can configure the time fields at the lower edge of the VIP window
There you have the possibility to edit the values manually by double clicking in the corresponding field. You can also use relative values in this fields, when you enter e.g. "+2:00" in the time fields, you increase the value of the field by 2 (seconds, beats...depending on the used unit of measurement).

You can also access this function by the keyboard shortcuts alt+ numpad 1..5 or with the corresponding menu items "menu Range→→edit time display→→ field 1..5".

1. By double-clicking in the relevant box.
2. By selecting the box via the Range > Edit Time display menu.
3. By using the shortcut keys (Alt+Num pad 1-5).

"CD/DVD" Menu

The options found in this menu are geared toward preparing and burning CD's.
If your computer is powerful (and well-tuned), you will easily be able to Burn Red Book Master CD's in real-time, as you mix a song or master another mixed track.

*Note: As new CD/DVD writing hardware is constantly appearing, MAGIX cannot guarantee that your hardware is currently supported by Samplitude's existing CD/DVD writer *.dll files. In addition, some CD/DVD writers have various hardware or firmware limitations. Therefore, prior to purchasing a CD/DVD writer for use with Samplitude, please download the most recent CD Burner DLL patch from the Samplitude/Samplitude Support Area (www.samplitude.com, navigate to "Support"). After installing it, you'll find a document CDR_Readme.txt in the installer folder, which is describing all compatibility-tested devices.*

If you have already purchased your CD/DVD writer, please do the following before using it with Samplitude:
1. Ensure you have installed the latest firmware update from the hardware manufacturer.
2. Download and install the latest CD writer *.dlls from *www.magix.com*

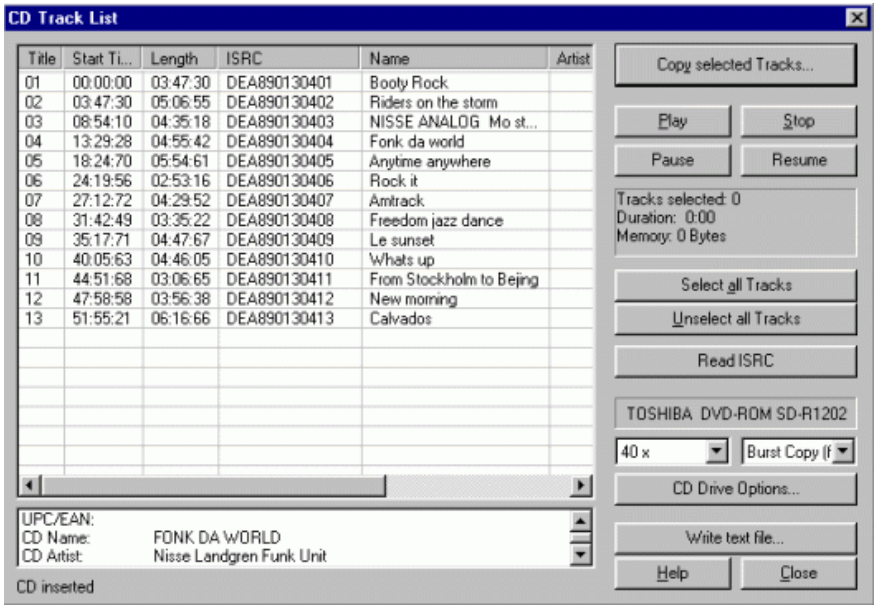
Samplitude allows you to create audio CDs directly from Virtual Projects and any stereo Wave Project, regardless of the number of tracks in the VIP.
32Bit Float/24bit Objects are dithered, and converted to 16bit, during the burning of the CD or the CD track bouncing function. The only requirement is that the project be at a sample rate of 44.1kHz.
When selecting the special "CD Arrange mode", regardless of the selected Range, Wave Projects are loaded by automatically inserting a pre-determined space between the songs, or Objects. The duration of this gap can be adjusted with the *CD > Set Pause Time* menu item.
If a completed, multi-track, VIP is destined for use as a single track on a (multi track) CD, use the Track Bouncing function to turn the multiple tracks into a single Wave Project.

Load Audio CD Track(s)

Import entire audio CDs or single CD tracks in a project. However, as opposed to normal data, Audio CDs have to be read first (grabbed or ripped). The data import is digital, so there are no losses in the sound quality.
If you want to import audio CD tracks, just follow these easy steps:
1. Insert an audio CD into your CD-ROM-drive and select "Load CD audio track" from the CD menu. A dialog with a list of the CD tracks will appear. If you have more than one CD-drive (or for example an additional CD-writer) you will first have to choose the drive from which the CD is loaded. You can do this in the CD-drive options

2. Select the desired title(s) (pressing Shift + mouse button or Alt + arrow keys).
3. Click on "Copy selected track(s)..."
4. The "Project import" dialog will appear. Here you can choose a file name and the target directory.
5. Now, the audio material will be copied from the CD-drive to the hard disk. A state display informs you of the current transfer state.
6. Once the audio data has been transferred, all dialogs will be closed and the tracks will be added to your master track as single Objects.

The Track List Dialog



All CD tracks are shown including start time and length. In the name column the track name is shown, this is derived from the CD Text information, if the CD provides this information and the CD drive is able to read it. Select a track by clicking it with the left mouse button, select a Range of tracks with SHIFT + mouse click, select several tracks with CTRL + mouse click.

Copy selected track(s): This button starts the audio copy process. For each track, a new Object and the corresponding Track Marker will be created in the arrangement.

Transport control: Use is like the transport buttons on a real CD player, start, pause and stop playback and skip forward and back between the tracks. Below this, information about the total length and memory demands of the selected tracks is displayed.

Select all tracks: all audio tracks are selected (for example, if you want to copy all tracks of a CD). Pressing Ctrl and clicking on the mouse button you can select more than one track.

Read ISRC: reads the ISRC code (if provided) of the tracks.

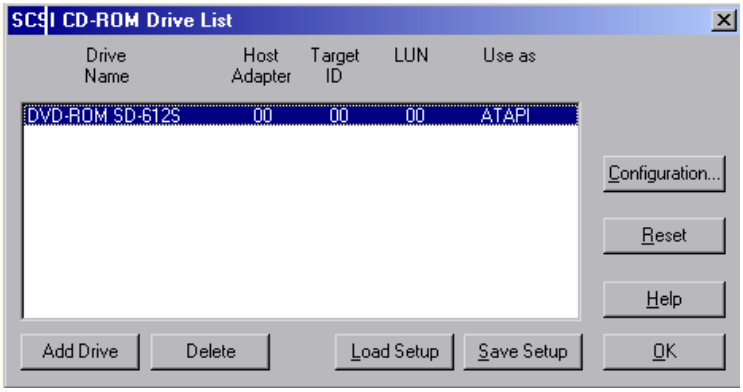
Tempo mode: In the right selection box you may choose the reading speed, in the left selection box the selection mode (see "The CD drive configuration dialog")
CD-drive options: Here you can adjust some of the settings of the CD drive settings you are using and choose the drive for "ripping" CDs if you have more than one CD-drive installed in your system (see The CD-drive dialog)
Write Text file: Creates a text file with the retrieved CD information as displayed in the CD track list

The "Import project" dialog

The "Import project" dialog appears after selecting the "Copy selected tracks" option. Type in a name and a target directory for the new audio files. The audio files will receive a number in addition to the name you chose ("name" -> name_1.wav, name_1.wav...).
The preset format for audio tracks (CDA files) is the WAV format. You are however free to convert them into a compressed format, like MP3 while still reading the data. Just choose your audio format in the "Import project" dialog and click on the "Format settings" button to make your changes (More information concerning the audio formats see Import Sample)

The CD-drive dialog

Choose and configure your CD-drive. The CD manager permits the import of audio data using most SCSI- and ATAPI-CD-ROM drives and CD/DVD writers. In case of doubt, ask our technical support which drives are compatible. If you experience problems while reading audio tracks, you can choose between several reading processes in the Configuration dialog, which differ in the way the data is read or "ripped".



Configuration: This button opens the Config dialog, in which you can set different special features, SCSI-IDs etc.

Reset: Restores the initial drive configuration.

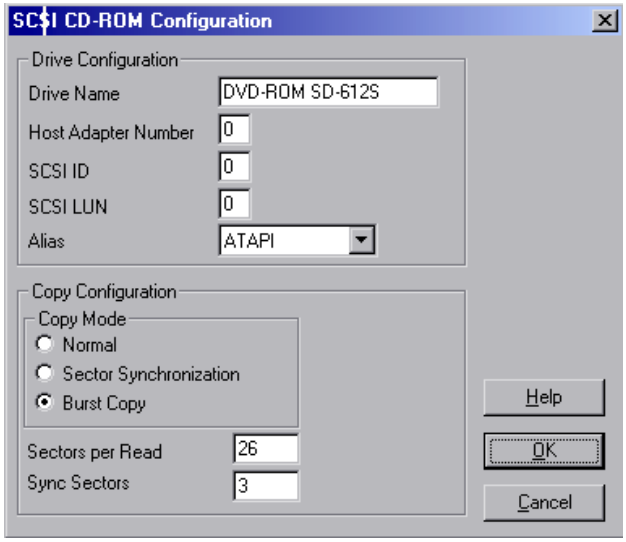
Add drive: Adds a new entry to the drive list. However, you will have to adjust the specific settings.

Delete: deletes a selected drive from the list.

Save setup: Saves the current drive list with all configuration-data in a *.cfg file.

Load setup: Loads the current drive list and all configuration-data from a *.cfg file.

CD-Drive Configuration



Drive Name: Lets you edit the name of the drive in the list. This is useful if you create more than one entry accessing the same physical drive.

Host Adapter Number: Lets you specify the number of your SCSI adapter - normally 0.

SCSI-ID: Lets you set the ID of your CD ROM drive. Be sure to set the correct ID, there is no error checking!

SCSI-LUN: Select the SCSI LUN parameter, normally 0.

Alias: Lets you select the manufacturer type of your CD ROM drive.

Copy mode Normal: Copies the audio data without any software correction.

Copy mode Sector Synchronization: Copies the audio data using a software correction algorithm. This is useful, because some CD ROM drives cannot seek exactly to the same position between two read accesses but Samplitude can correct these differences using this algorithm.

Copy mode Burst: Optimizes the speed of the copy process, no software correction is done.

Sectors per Read: Defines the number of audio sectors per read cycle, the higher the number the faster the copy process will be. Not all SCSI adapters support more than 27 sectors!

Sync Sectors: Defines the number of audio sectors used for the Sector Synchronization. A higher number results in a better synchronization but also in a slower copying process.

Set CD Track Index

Choose this option to place a Track Marker at the current play cursor position. all subsequent Track Markers will be re-numbered.

Important note: Each audio track on the CD MUST have a Track Marker! Track Markers are placed at the beginning of each audio track (slightly before the audio starts). The Track Markers allow CD players to cue to each individual audio track.
Tip: This option is also available from the toolbar.

Set CD Sub index

Choose this option to place a Sub Index Marker at the current play cursor position. all subsequent Sub Index Markers will be re-numbered.

Note: Sub Index Markers are not required, but they allow CD players to cue to specific points within an audio track. Sample CD's containing dozens or hundreds of loops, broken into components, often use sub indexes.
Tip: This option is also available from the toolbar.

Set CD Pause index

Choose this option to place a Pause Marker at the current play cursor position.

Note: Pause markers allow CD players to switch their output to absolute silence until the next Track Marker.

Set CD end index

Marks the end of the CD. This is important for burning a CD, when the project still produces audio after the endpoint of the last Object(s) e.g. a reverb tail. To prevent Samplitude from cutting this section of audio, you can place the CD End Marker an appropriate distance after the last Object. Another use of the CD End Marker is to partially burn a project to CD. CD Burning always starts from the first Track Marker, and ends with the End Marker. all you need to do is delete all Track Markers that precede the first track you want to burn. All audio before the first Track Marker will not be burned to the CD. Burning stops with the CD End Marker, allowing you to place the CD End Marker inside the project.

Set Track Indices on Silence

This function sets Track Markers on silent passages of a selected Object.
As an example, this is useful for setting markers between each title, after recording an entire DAT tape.

Set Track Indices on Object Edges

Choose this option to automatically place a Track Marker at the beginning of each Object (in track 1). In many situations, this option provides the quickest way to set Track Markers for your CD.
Prior to using the function, execute the "Remove all indices" command, to delete any redundant Track Markers.
If there are multiple Objects that make up a single track or title, you may want to first use the bouncing function (to combine the Objects), to ensure proper track assignments.

Tip: This option is also available from the toolbar.

Set track indices on Object edges – Options

Set Pause indices on Object ends

This option automatically generates Pause markers on Object ends.

Time Offset for indices on Object Edges

This option allows you to set an offset for the generated Track Markers.

No indices on Object Crossfades

This option prevents track indices from being placed between two overlapping Objects, which segue via a crossfade. This makes it possible to use the automatic Track Marker generation functions, even if crossfades are used inside each track.

Remove index

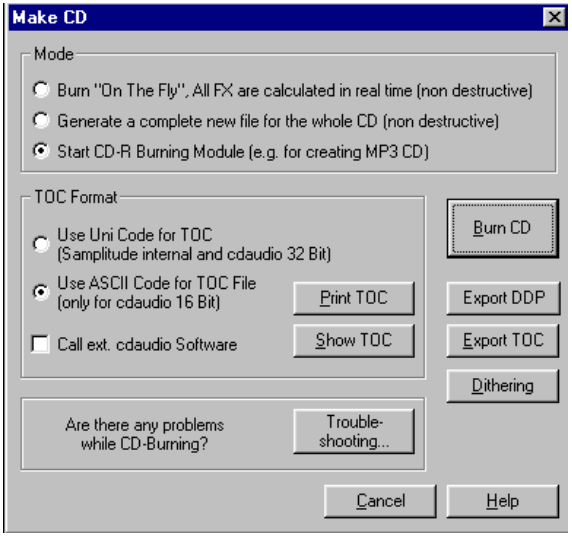
Choose this option to delete a Track Marker or Sub Index Marker. First, click on the marker, then select this option to delete it.

Tip: After clicking on a Track Marker or Sub Index Marker, you can press the delete key to remove it.

Remove all indices

Choose this option to delete all Track Markers and Sub Index Markers. This can be helpful before use of the "Set track indices on Object edges" function.

Make CD...



This dialog starts the CD writing process. Samplitude contains high-grade CD creation routines that are constantly updated. The software code is licensed from Point Software & Systems.
If you want to find out which CD-R drives are supported, please read the file "CDR_Readme.txt" in the installation folder or visit our web sites at www.samplitude.com.
Samplitude creates a TOC (Table of Contents) file prior to starting the CD creation. The name of the current VIP is used and the extension *.tcx is added to the file. The TOC file is located in the same folder as the current VIP. It is therefore important to save the VIP to the hard disk prior to starting the burning process.

Burn "On The Fly": This mode will burn the CD directly from the VIP window. All real-time processing will be calculated (included) during the burning process, so there is no need to create an Image File. This saves a great deal of time and hard disk space!
The following real-time tasks are calculated during the process:
– Object effects, volume and panorama settings
– Fades and crossfades
– Mixing of tracks
– Mixer track effects
– Mixer master section effects

– 32bit float to 16bit conversion and dithering

To get a good idea whether your system is able to sustain the real-time processing needed for this functionality, try this:
Play back the VIP sections that contain the largest number of tracks and/or where the most real-time effects are used. Watch the DSP meter (bottom left-hand). Below are the expected performances at the different DSP values:
– Below 25%: CD creation with up to 4x mode
– Below 50%: CD creation with up to 2x mode
– Below 90%: CD creation with up to 1x mode
– Above 90%: Real-time creation is not possible, use the second mode (Bouncing)

Generate a complete new file: Use this option if your system is not fast enough to sustain the real-time creation of the CD (see above). This mode calculates all WAV files, including fades, crossfades, and volume automation into a new audio file. Any other real-time processing is also part of the newly created audio file. Make sure that you have sufficient hard disk space prior to starting the process (approx. 700 MB for a complete CD).
The original WAV files used in the VIP remain unchanged. This makes this procedure non-destructive!

Use Uni Code for TOC: This is the default setting and is used for the internal CD creation process as well as the 32bit version of Point CDAudio.

Use ASCII Code for TOC: This format should only be chosen when using the 16bit version of Point CDAudio.

Call external CDAudio Software: This lets you determine whether you want to automatically start the CD creation application "CDAudio". The current TOC information is passed to the application, so that the CD creation in the external program can be immediately started. all Samplitude projects are closed and the application itself is closed down to prevent any type of file access conflicts.
If CD-Audio is installed in any other folder than "C:\CDAUDIO" you will need to enter the location of the application.
Only use this option if you do not want to use Samplitude's internal audio CD creation and purposely want to use an external program.

Start CD-R burning module: starts the MAGIX CD burning utility to burn any data files, e.g. mp3 files on CD.

Print TOC: This starts the external TOC Printer application. This convenient tool allows you to print the contents information of the current CD. You may choose between a text style format to print the production documentation and a formatted printout for the CD jewel case. Please refer to the on-line help for the TOC printer for more specific information.

Show TOC: This button opens a text window, which shows the contents of the current TOC. The "Copy" function can be used to copy the contents to the Windows clipboard for use with other text editing applications.

Export TOC: This function allows you to save a TOC file (*.TOC, *.TCX). A track-bounced Wave Project is also created from the VIP, which is referred by the TOC file.

Troubleshooting: Opens the CD Burning Profiler, which checks the system for incompatible drivers of your CD-Recording hardware.

Create DVD-Audio

Introduction

Samplitude can burn DVD-Audio Discs using any type of DVD writer. You can burn the following disc formats: +R/-R/+RW/-RW.
DVD-Audio discs can only be played back on DVD-Audio enabled players! DVD-Audio-enabled players are identified by the DVD-Audio logo on the device. The player requires up to six individual analog outputs for multi-channel playback.
Samplitude currently burns so-called "Black Discs", DVDs without a graphic menu but merely including Track Markers which can be controlled like an Audio CD. However, to create DVD audio/video with menus, videos and slideshows, the video cutting and DVD authoring program "MAGIX movie edit pro" will soon be available which can be integrated into Samplitude.
You can save 16 and 24 bit audio on a DVD-Audio; sample rates of 44.1 and 48kHz as well as double and fourfold amounts of these rates are possible. Up to 6 audio channels are supported in 5.1. surround as well as other configurations (stereo, 4.0, etc).
The only limitation is that the maximum allowed bandwidth for hardware players of approx. 10Mbit/s must not be exceeded. Therefore, only sample rates of 48kHz are available for 5.1 surround sound. The 10Mbit limit would be exceeded at 96kHz.
For example, the following playback times would result for the available storage capacity of a single-layer DVD-R (at 44.1kHz sample rate):
Stereo 16 Bit: approx. 7h
Stereo 24 Bit: approx. 4.5h
5.1 surround and 24 Bit: approx. 1.5h

Creating a DVD-Audio disc

In the VIP, Track Markers are placed in the same manner as for a CD. An Audio CD and a DVD-A can be subsequently burned from the same project.
{bmc dvda_button.bmp} start DVD-Audio creation via the "DVD-Audio" button in the CD toolbar.
The "Trackbouncing" dialog is displayed. You can choose between 16bit and 24bit audio. The initial setting is 24bit. The "Create DVD-Audio" box must be activated.

An external program is opened for burning the DVD-Audio. The path to this program is predefined but can be changed if necessary.
Alternatively, "Movie Edit Pro" can also be opened here, which enables advanced DVD authoring (menus).
"OK" starts the bouncing process. The DVD-A burning application is launched and the files are handed over automatically.
Note: Double storage space of a DVD is required on the hard drive, as the VIP is bounced first and the DVD image is then created from the complete audio data, which is then burned.
The track list is displayed in the DVD audio tool. Further tracks can be added manually, and the surround format can be changed (stereo, 4.0, 5.0, 5.1).
The "Burn" button opens the burning window where the burner and burning speed are selected.
The DVD image is first created after pressing the "Start" button. This is then followed by the burning process.

Show CDR drive Information....

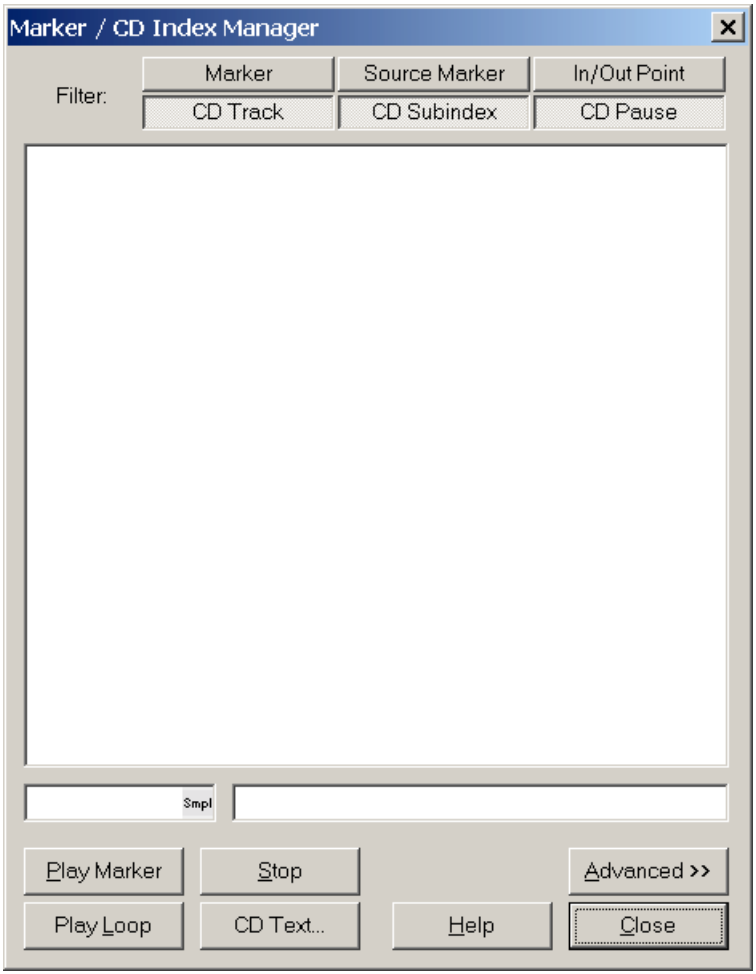
Choose this option to display information about your CD-R drive. *A CD must be in the drive for this option to work.*
Among the information displayed is the manufacturer, drive name, hardware revision, cache size, and the features supported by the drive mechanism.
The "Disc At Once" feature is extremely important, as it is needed for Red Book-compatible Audio CD production. CDs burned in this way are accepted as Masters by CD Manufacturing facilities.

Show CDR Disc Information...

Choose this option to display information about the CD currently in the CD-R drive.
The most important information is the maximum length. This cannot be exceeded during the production (i.e. 74 minutes and 5 seconds).

CD Track/Index Manager

In this dialog, all the CD tracks and sub-indices in the current VIP are displayed in a list.



All currently saved markers of the active project are shown with their name and position in this dialog. Select markers with the mouse, then you can change its position Normally in the time entry field below the list. You can also name and rename it there. Play marker starts playback up from the selected marker, Play loop plays a one second loop around the marker. Stop stops the playback. CD Text opens a separate dialog for entering CD Text information.
The play cursor is moved to the respective marker position by double-clicking on the desired marker.
With Filter you can select, which of the marker types to display.
Create: with the according buttons, you can create markers, CD tracks, CD Sub indices or CD Pause markers at certain time positions. You have to enter the time position in the time field below the marker list.
Other settings such as Preemphasis, Copy Protection, and Second Generation Protection Flags can be set for each of the tracks.

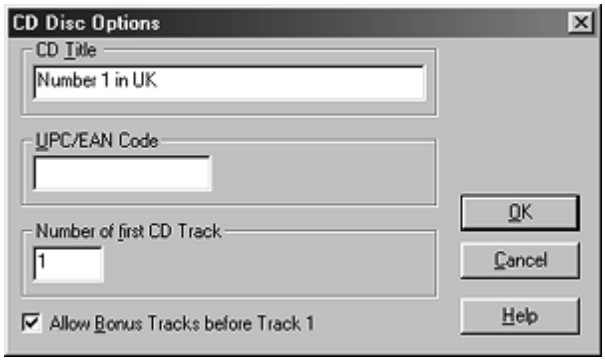
The button Apply to all allows you to use the current flags to set all tracks to the chosen settings.
Other settings are the ISRC Codes for every CD track.

CD Text/MP3 ID Editor

Here you can input the CD text information for the CD. The CD track names are copied from the CD Track Marker names in the VIP. all information here is contained also in the VIP, but in addition to that you have the possibility to save and load all entries made here in a separate file (*.cdt).
This function allows you to supply data regarding a song or CD content – such as album, artist etc.
MP3 files do not only transport audio data, but also information about the coded music piece through so-called "ID3 tags". These are file appendages into which an encoder can write standardized information. The ID3 tags are recognized by the decoders and displayed as music data by the MP3 player.

CD Disc Options....

This option opens the CD Disk Options window, allowing you to configure several parameters for the CD.



CD Title: This line allows you to enter a Title for the CD. The Title is written to the CD, and is used when printing the TOC.

UPC/EAN Code: This line allows you to enter a UPC/EAN Code for the CD. The UPC/EAN Code is written to the CD, and can be requested by special CD players.

Number of first CD Track: Under certain circumstances (e.g. Track-At-Once burning), the number of the first track can be determined with this setting. As Samplitude burns Red Book Audio CD's in one pass (*Disc At Once)*, you will probably never need to adjust this setting. (It has no bearing when Burning Disk at Once.)

Allow Bonus Tracks before Track 1: This option allows you to insert "hidden" or bonus tracks before Track 1.

Set Pause Time....

Choose this option to set the default Pause Time between tracks. The amount of time can be entered in Samples, Milliseconds, SMPTE, or bars:Beats:Pulse formats.
The Pause length is required for the grid function, allowing sample Objects to ”snap” to the edges of other Objects + the Pause time. A Pause time of 2 seconds is normally used.

Note: The SMPTE frame rate can be set in the View > Units Of Measurement menu. The bars/Beats format can be defined by the View > snap and grid setup > Bar Definitions options.

Set start Pause Time....

Choose this option to set the default Pause Time at the start of the CD. The amount of time can be entered in CD frames (75 frames = 1 second). *Important!!! This setting must be set to a value of 150 frames (2 seconds) to meet Red Book audio CD requirements!* It is recommended that you set this parameter to 150 frames (two seconds) and leave it there.

CD Arrange mode

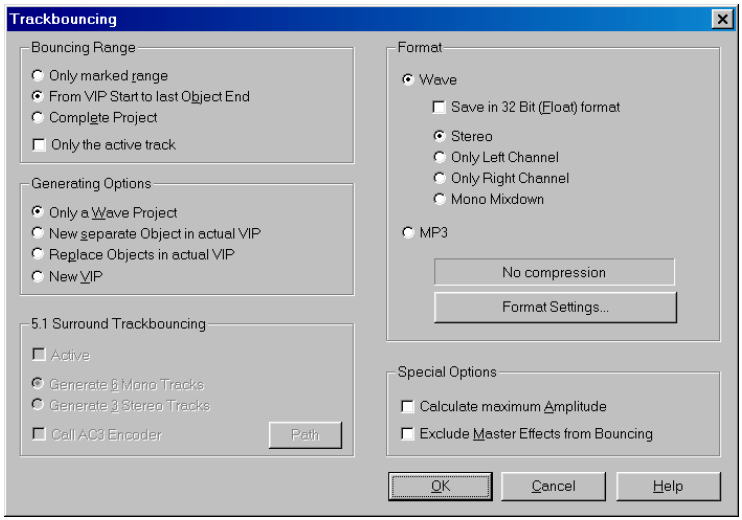
When you activate this menu item, Samplitude rearranges newly inserted Objects, and inserts pauses (conforming to the Red Book Standard) between the Objects.
We recommend the following procedure:
- Open a new VIP which contains 4 tracks so you will be able to mix down later, if required.
- Activate the Special CD Arrange mode via the CD menu.
- Open a new VIP.
- Load audio files, audio tracks, or make a recording using a microphone.

You will see gaps between the individual Objects in the VIP – these are the inserted pauses. The length of the inserted Pause(s) can be defined in the ”Set Pause Time” dialog.

"Tools" Menu

Track Bouncing....

This option opens the Track bouncing dialog.



This window allows you to "Track bounce" a Virtual Project (VIP) to one or several Stereo audio files. This is essentially the same as exporting the VIP via the Export Project command in the File menu, but offers many more possibilities. You can also export single or multiple Objects or tracks, restrict the export process to Ranges and exclude Track and Master Effects. The resulting file can then automatically be opened in a new VIP or replace the Objects in the current one. All tracks and all real-time processing (fades, effects, volume and pan automation, Object effects, etc.) will be included in the bounce. If you are bouncing to 32 float format, no distortion can occur.
It makes sense to use the master normalizer function of the mixer to create a file with exact attenuation.

Trackbouncing Settings
Source:

Only marked Range: Choose this option to limit the bounce to the left and right boundaries of the currently marked Range.

Bouncing is performed from the project beginning to the end of the final Object with reverb decay (refer to Program Options).

Complete Project: Choose this option to bounce the entire project.

Note: When selecting this option, ensure that no (or little) blank space exists at the end of the VIP (i.e. after the endpoint of the last Object). Blank space will be included

in the bounce! This can result in very long bounce processing times. You can easily delete the blank space by selecting it as a Range, and pressing the delete key.

Bounce selected Objects: all selected Objects are bounced, i.e. all Object effects are applied and Wave Projects are created, one for each single Object. The Objects are replaced by those referring to the new Wave Projects. This permits quick export of a large number of single Objects to separate files (e.g. for sample archives).

Glue selected: All selected Objects of a track are included in a new file with all Object effects.
These options has the same function as the "Glue Objects" function of the "Object" menu; you can enter a custom file name, but there is no "Freeze/ Unfreeze" mechanism.

Options
Process only marked tracks: Only the selected track is bounced. If "Only marked Range" is also activated, the marked Range is only bounced in the marked tracks.
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 groups.

Multi-track bouncing: bounces each track (including auxes) to a separate file, for further processing in an external multitrack environment.

Show maximum Amplitude: Click on (enable) this option and (when the bounce process has finished) Samplitude will display the maximum amplitude reached by the "bounced" audio file.

Effects:
Exclude Master Effects from Bouncing: The master effects are not included during bouncing.

Exclude Track Effects of the Target Track: The track effects of the target track (if not identical with a source track) are not included.

These options are important in combination with "New separate Object in current VIP/Replace Objects in current VIP" so that the track effects are not applied twice. They are automatically adjusted when selecting a different "Generating" option.

Generating
Only a Wave Project: The bounced file is saved as a Wave Project. No new VIP will be generated for the bounced file.

New separate Object in current VIP: The bounced file is saved as a Wave Project, and a new Object which references this file is created on a separate track in the active VIP.

Replace Objects in current VIP: The bounced file is saved as a Wave Project, and a new Object which references this file is created by replacing the original Object(s) in the active VIP.

New VIP: The bounced file is saved as a Wave Project, and a new Object which references this file is created and placed in a new VIP.

Note: In all cases, the original Wave Projects are not changed. That said, however, as some of the options result in changes to the VIP, we recommend that you save and backup the VIP before using the command.

Format

The target format can be set flexibly. WAV and AIFF formats are supported in 8/16/24/32 bit mono/stereo at variable sample rates, mp3 at variable bit rates, and WAV with any installed ACM codecs.

Note: All tracks that are the result of a trackbounce are opened (even if merely the "Only a Wave Project" Generating option has been selected). This means:

1. You cannot bounce the same project twice in a row if it is still opened as a result of the first bouncing process. It must be closed first. If you often bounce and no longer require special options (as for mastering), you will be faster if you select àà Export Audio from the "File" menu instead of using the trackbouncing window.
2. All formats that cannot be directly opened by Samplitude cannot be created in the trackbouncing window. For example, to bounce into Real or WMA format you should also use File àà Export Audio.

Presets

You can save all settings in this dialog into presets. So you can restore your special settings for, e.g. generating mp3 preview, Multi Track Bouncing or Object.

Surround Track Bouncing:

This enables the surround trackbounce functionality. The other options in this section will not be available until the "Active" box is checked. This is only possible if the Project is set up for a surround mix. Read the according section about Surround Trackbouncing in the Surround chapter.

OK

Click on this button to start the Track Bouncing process.

Cancel

Click on this button to close the Track Bouncing window, without performing a bounce.

Bouncing Sub Mix Busses

You can also bounce a Sub Mix bus. Every track routed to the Sub Mix bus will be included in the bouncing process, even if the track sending to the bus is a bus itself.
This is handy when you want to mixdown all tracks sent to a specific bus (grouped strings, a drum kit etc.).
If you bounce several tracks, or just the active track (Option: only marked tracks/active track) with the "New Object in actual VIP" or "Replace Objects" functions - and this selection includes a Sub Mix bus - then it is possible that the resulting Object will be placed at just this Sub Mix bus. It's actually pretty likely, in fact, because busses usually don't contain Objects (although they can), and the Object will be placed in the first empty track it finds. In this situation, the track FX (including pan/volume automation) of the Sub Mix bus will be excluded from the bouncing process. This prevents the effects from being applied twice (once by the bounce, and once from the real-time track FX).
You cannot bounce aux busses in this way. You need to include the tracks being sent to these busses, in order to obtain the desired result.

Range Bouncing

Use this function to convert the Objects within a selected Range to a new WAV file. The selected Objects in the VIP are replaced with the track-bounced version. This is very useful to combine multiple Objects into a single Object so that the function to create Track Markers automatically can be used.
If the VIP contains 24bit or 32bit Objects, a dialog box will ask you whether the new file will be created as a 32bit float or 16bit file.

Remove unused Samples

This function removes all material from Wave Projects belonging to the current VIP that is not used by any of the VIP Objects. The Objects in the VIP will point to the proper audio Ranges in the linked audio files. This leaves the VIP itself unchanged.
This function physically deletes data and has no undo capability, so use it with caution!
If multiple VIPs use the same Wave Project, all VIPs accessing this projects have to be open to avoid data loss.
Using this function can conserve a lot of storage space. However, the downside of using this option is that after performing this function, Objects can only be made shorter, but they cannot be extended any more, because all audio data outside the Object's borders was removed. For this reason you can also define a security Range in samples. This amount of samples will be left in the Wave Project before and after the according Object border to have some reserve when

you want to change a fade in/out of the referring Object. The default value is 22050 (= 500ms at 44.1 kHz sample rate).
The dialog lists all Wave Projects referenced by the VIP and the amount of storage space, which the Wave Project consume in total and how much of it are actually unused in the VIP. Every file has a checkbox, to include it in the process or not. Only files with unused samples are checked by default, you can uncheck any Wave Projects you want to keep completely.

Note that the list also contains files not actually referenced by the VIP, but referenced by the VIP's undo chain. These files are 100% unused and will be deleted completely when checked!

This can be right (when you have a recording session rejected), but there are also cases, when you just imported audio material from other sessions or your private sample library and decided later, rather not to use it.

Tip: If you need the complete contents of the audio files for other production projects, it is recommended to archive the finished project in the following way. Save the complete project to a new folder (File->Save Complete Project to). The VIP is now located with the complete contents, in other words all the audio files, in an unaltered form (WAV, RAP, etc.) in the same folder that the VIP was saved to. Next, use the command Remove Unused Samples. The same folder will now only contain the audio data or samples that are actually used by the archived VIP. The contents can be easily back up to any backup medium such as CD-R, Data DAT, and others.

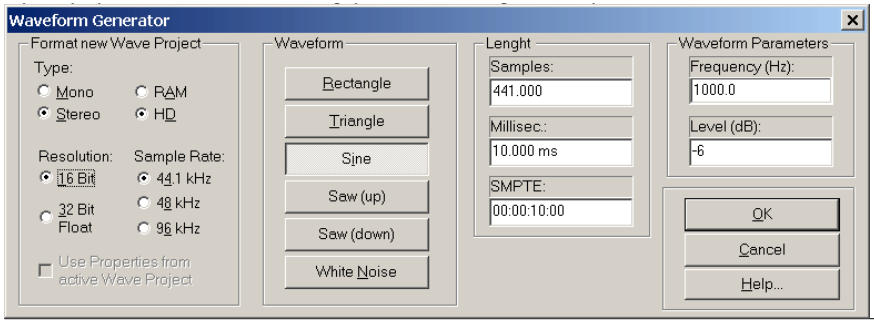
Delete Freeze Data

Please refer to the chapter "Freeze"

Collect Project Files

If a project uses audio data from different sources, sample CDs or audio data from other projects these files often reside in different directories. It is wise to collect all project-relevant files into the project directory to make sure nothing is missing when you backup the project.
You can copy or move the file. When moving, make sure the file is not used by another project.

Waveform Generator



The generated waveform shares the properties of the opened Wave Project. As an option, the test tone can be automatically loaded into the active VIP.

Format new Wave Project

These options allow you to select the type, resolution and sample rate of the new Wave Project that will contain the generated tone.

Waveform

Six waveform varieties can be selected – square, pulse, triangle, sine, saw up, saw down and white noise.

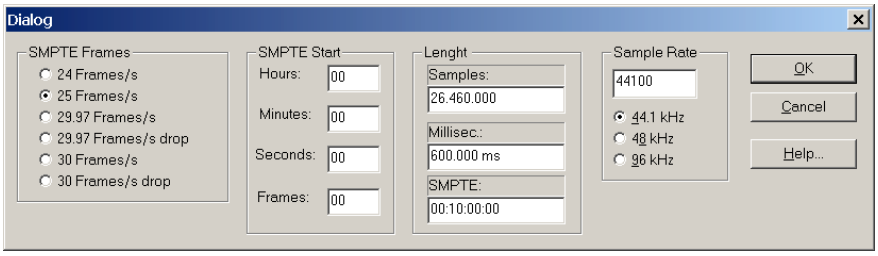
Length

Use these boxes to enter the desired length of the generated tone.

Waveform Parameters

Use these boxes to set the frequency and amplitude of the generated tone.

SMPTE Generator



This dialog allows you to create audio files containing SMPTE time code. These files can then be placed into VIP tracks, and routed through any audio output of your PC. This ensures extremely stable SMPTE time code output, which is sample-locked to the audio material in your VIP.

New Manager...

This command opens a further Manager window. The Manager key on the key bar shows/hides all managers.
The Manager window has several different views that represent the different managers.

Please refer to the chapter "The Managers"

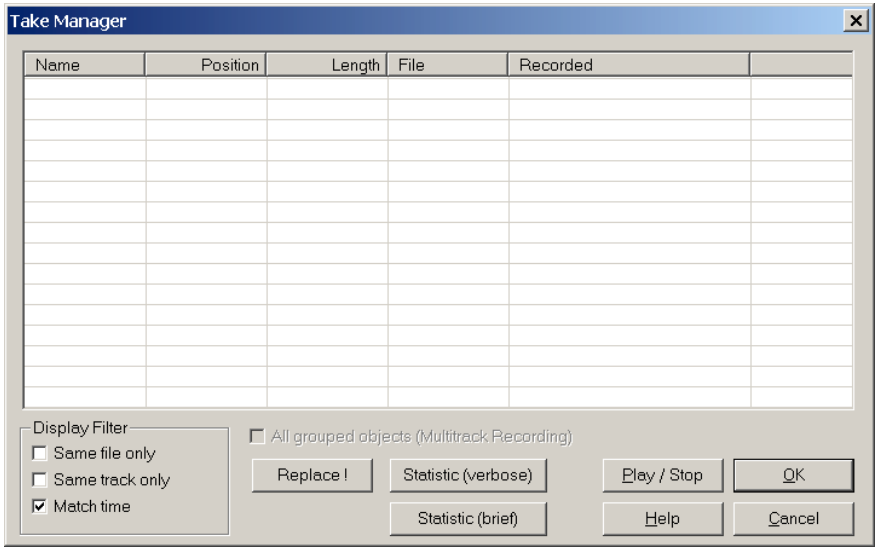
Manager → File Browser/Object Manager/Track Manager/ Marker Manager/Range Manager

Please refer to the chapter "The Managers"

Take Manager...

Opens the Take Manager window. The Take Manager is specifically designed for users who work with large numbers of different recording takes.

Samplitude registers the VIP position of the recorded audio material. This "time stamp" is stored in the HD or RAM Wave Project, and offers the option to return the recorded Object to its original position.



Display Filter

Same file only: Only takes that belong to the same audio file as the selected Object are displayed.

Same Track only: Only takes that are positioned in the same audio track as the selected Object are displayed.

Match Time: Only takes that match the original time position of the selected Object are displayed.

Play

Starts playback. Please set the play cursor to the desired playback position, before opening the Take Manager.

Replace!

This replaces the selected Object with the take(s) currently selected in the Take List.

Statistic (verbose)

Creates a new VIP with the currently visible takes in the Take List. Every take is listed in a new track. The VIP can be used to audition the individual takes, by making use of the solo and mute functions of each track. This facility can be useful in situations where takes are confusingly named, during high pressure (fast) sessions.

Statistic (brief)

Creates a new VIP, with the currently selected takes in the Take List. All takes are aligned on the same track.
This facility can be useful for burning a CD of the takes for later listening, or general archiving purposes.

Take List

The Take List shows the Objects, allowing you to search and match other Objects with the selected Object. The list shows the take name, time position, the Wave Project associated with the take, and the date/time stamp.

Note: You will thank yourself later if you spend the time in appropriately naming takes and Wave Projects, during recording sessions. We understand that the creative flow occasionally dictates that this is not possible, but the more you practice, the faster you will become.

Using the Take Manager

For this example, lets assume you have an Object in the VIP that was processed (non-destructively – "make copy" option enabled) with the Room-Simulator. In this situation, the processed (wet) signal was appended to the end of the original (dry) recording. After some critical listening, you decide that the Object sounded better dry, and now wish to swap the wet Object, for the original dry take (Object).
Here's how it's done:

1. Select the Object that you'd like to replace (swap) by clicking on it (in the lower half of the track).
2. Mark a Range that encompasses the currently selected Object. This is to ensure that playback occurs in the desired area, while auditioning changes.
3. Open the Take Manager window (*Tools > Take Manager*, or *Object > Take Manager*)
4. In the lower left corner of the Take Manager window (under Display Filter), ensure that "Same Track only" is the ONLY enabled option.
5. Look at the Take List, you should now see two takes. The take featuring the red "O" is the currently selected Object.
6. Click on the name of the original (dry) take (the one *not* marked with a red "O"), and press the *Replace!* Button. Samplitude will now replace the wet Object (take) with the original dry Object (take).
7. Click on the *PLAY* button to audition the change made in step 6.
8. Click on the *PLAY* button a second time to stop playback.
9. Click on the *OK* button to accept the change you've made, and to close the Take Manager.

Note: Samplitude "Time Stamps" each Wave Project it records. This allows the Object (representing the Wave Project) to be returned to its original start position, if accidentally moved.

The first step in working with the Take Manager is to select an Object. This could be the most recently created Object, following a punch-in recording, for example. This (selected) Object is identified with a colored "O" in the Take list.

Note: The Take Manager window will not work on Objects that were not recorded by Samplitude! As an example, .WAV files imported via the File > Import Sample option, or CD tracks loaded via the CD > Load Audio CD Track(s) option. If you select an Object that was not recorded in Samplitude, and choose the Object > Set new original position option, Samplitude will time stamp the Object, allowing it to be used in the Take Manager.

Some examples of Take Manager use may be:
– Selection of the best Take, following several Loop-punch recording passes.
– Locating the best material between bars 32 and 34, from 12 Takes of a classical production.
– A clear overview of all available Takes at a specific SMPTE time index (e.g. 30:00 to 35:00).

Start Windows Explorer

This opens the Windows Explorer. If you position the Explorer window to the left or right of the Samplitude window, you can easily drag & drop WAV files into the Samplitude projects. This allows easy management of numerous files (such as sound effects). This even works with dragging of multiple files.
Any file type with the WAV, OBJ (for Objects), HDP, and RAP extension can be dragged into the VIP.
It also works during playback.
This is an obsolete command left in Samplitude for those who are used to use it. You can browse your hard disc much more comfortably with the file browser from the Managers. It includes a text search function and favorite directories.

Timestretch/Pitchshift Patcher

Overview

This tool enables audio files used in Samplitude and other MAGIX programs (e.g. MAGIX music maker) to be patched with certain meta data that contain information about time and pitch of audio ranges. Additional information and settings are written into the Wave Project thus ensuring better timestretching or pitchshifting results.
The following can be patched:
– Algorithms for Timestretching/Pitchshifting
– BPMs
– Beat markers

Algorithms for Timestretching/Pitchshifting

Different algorithms (besides the standard algorithm) can be used for real-time timestretching and/or Pitchshifting for better results. Check Resampling/Timestretching/Pitchshifting for more details. With the patcher you can store

the best timestretching algorithm in the Wave Project. This ensures that the "correct" algorithm is used automatically when Timestretching/Pitchshifting is used on these projects as Object effects (e.g. with the Timestretch mouse mode).

Beat marker

Beat markers are also saved in the Wave Project when using beat marker-based algorithms.
In contrast to the Timestretching dialog, the patcher is non-modal, meaning that if the patcher is open, you can move beat markers in a Wave Project while checking out the Timestretching result.

BPM value

Patches the BPM value (beats per minute). This is useful for when the Timestretching factor must be determined later in order to adapt the Wave Project to the tempo of an existing arrangement.

Patches in brief

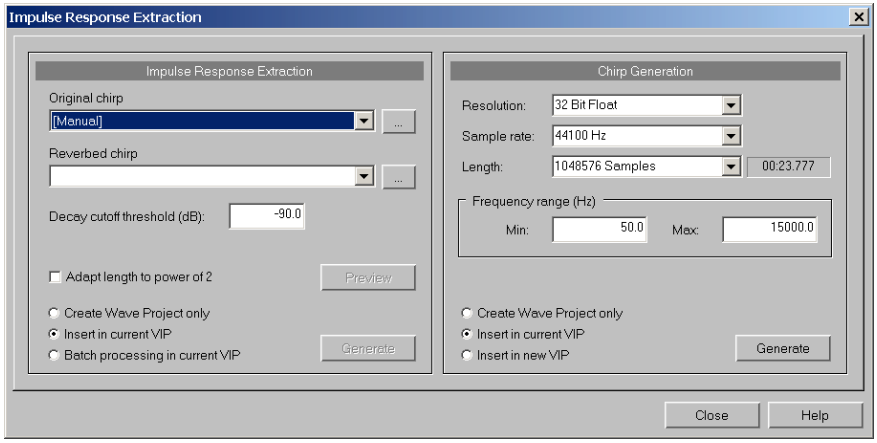
Patches only work if you have opened the Wave Project in destructive Wave Editing mode and if it is not open in a VIP simultaneously.
On the right of the dialog (Patch) you can select the algorithm to be used and specify the audio file tempo. On the left side (Test) is the control element from the Timestretching/Pitchshifting Dialog.
It allows you to test the selected algorithm in connection with the Play/Stop and Play original buttons. Patch saves the information in the wave file.
Samples from the MAGIX music maker product range Soundpool contain the patch information as of version MAGIX music maker 2003.

Remix Agent

Please refer to the chapter "Effects and effect plug-ins"!

Impulse response extraction

The "chirp" method for extraction of impulse responses is more efficiently than other known methods. It distinguishes itself, besides other advantages, by a very high robustness against long-time disturbances and non-linear distortion.
Impulse responses are used in the room simulator for the creation of ultra-realistic, natural sounding reverb.



General workflow

1. Generating a chirp.
2. Capturing of the room's "answer" to the chirp. Usually, that is playing the chirp in an environment which is intended to be "cloned" with a suitable PA speaker system and simultaneously recording the resulting sound of the room with a good microphone. But that can also be achieved by processing the chirp in an effect unit.
3. Calculation of the impulse response from the original and the reverberated or otherwise processed chirp.

Step 1 Generating a chirp

Open the "Impulse response extraction" dialog from the "Tools" menu. on the right side of the dialog there are the parameters for impulse creation.
A chirp is a sine wave with a continuously changing pitch.

Parameters:
Resolution: Choose between 16 bits and 32 bits float. 32 bit is recommended.

Length: Set the length of the chirp in samples. (always a 2nd number)
The longer the chirp is, the better the results are. The influence of continuous disturbances like hiss or traffic noise or non-linear distortion lowers with a rising chirp length. At the other hand the amount of calculations rises exponentially. Attention! The chirp should be at least as long as the reverb length!

Sample rate: Set the sample rate here. The sample rate of the original chirp, the reverberated chirp and the VIP (if it is used) have to be equal!
If you are working with 96 kHz sample rate and intend to extract natural impulse responses of room, so keep in mind that the speakers used for playback have to be able to reproduce very high frequencies in order to include the (very

small) ultrasonic components of the impulse response. Frequency Range: Set the start and end frequency of the chirp.
The frequency range should be adjusted according to the abilities of the used speaker. Avoid sub- base frequencies if the used speaker is not capable of reproducing these frequencies.

Options
Only create file: The chirp is just written into a new Wave Project.

Insert in current project: The Wave Project with the chirp is inserted into the current VIP.

Insert in new project: The Wave Project with the chirp is inserted into a new VIP. So you can play back the chirp and record the room response from this VIP at the same time.

Generate: Make the chirp!

Step 2: Reverberation of the chirp
Natural rooms
The chirp is played back monophonic with a speaker or speaker combination in a room. The speaker should be set up at that place where usually the acoustic sources (instruments, singers) in the room are situated - in a theater building that is the stage, for instance.
Speakers have a different directional behavior as typical natural acoustic sources, they send less sonic energy to the back. That's why the application of speaker combinations is recommended.
An often used speaker combination is to use one speaker in front direction and a second one in back direction.

Digital reverberation units
Reverberate (process) the chirp signal with the reverb unit.
The "Dry" part of the output signal should always be zero, otherwise the result will become unusable.

Important: The reverb tail has to be recorded until its very end! Avoid any impulse-like disturbances as well while the reverb fades out.

Hints for playback while recording the chirps in Samplitude (for recent version 7.0)
- In the dialog recording options (Key R) please choose "Playback while recording" and the Monitoring mode "Peakmeter Monitoring" because "Software Monitoring" can lead to acoustic feedback.
- In the dialog: activate "Monitor" and "Visualization".
- Start a test recording with "Record" and play back the complete chirp to adjust the recording level to an appropriate value. Avoid clipping by all means! Reject the recording and repeat the procedure until you find the optimal level.

– Open another visualization window with a spectrogram view and a level range from -90 dB to check the reverberated chirp for non-linear distortion, disturbing impulses or aliasing (details see below).

Step 3: Calculation of the impulse response from the original and the reverberated chirp

Source of chirp parameters: Choose the Wave Project or Object which contains the original (un-reverberated) chirp.
If the original chirp is not available anymore, you can manually adjust the chirp parameters on the right side of the dialog. Keep in mind that you have to set it exactly to the same values used for creating the reverberated chirp or it will not work at all.

Reverberated Signal: Choose the Wave Project or Object which contains the reverberated chirp. When working in a VIP (see below) this field is filled automatically.

Decay cutoff threshold: If the signal level falls below this value (in dB), it will be cut off.

Adapt length to power of two: Fade out the impulse response in a way that the length of the sample is almost a power of two (in samples). Use this to optimize the performance of the room simulator using this impulse response.

Option "Only create file": The resulting sample will be only written into a Wave Project.

Option "Insert into actual VIP": An Object with the impulse response will be created in the current VIP.

Option "Batch processing in current VIP": This works only when the reverberated chirps are all Objects in the actual VIP (read details below). With this option activated, impulse responses are calculated from all Objects in the list of reverberated chirps (supposed you use the same chirp for all). According to the list entries, new Objects with impulse responses appear behind the Objects with the reverberated chirps.
Use the Mute buttons to exclude single Objects from the impulse response calculation.

Hear: The impulse response is being calculated and played back.

Save: The impulse response is being calculated and saved.

Processing impulse responses automatically in a VIP

A typical and recommended workflow would be to place Objects containing different reverberated chirps below the original chirp's Object (typically on track 1) track by track.

Arranged this way, these Objects now appear in the list of reverberated chirps. The track number supplements the Object name for a better orientation. With the mute and solo buttons of the tracks you can exclude single Objects from appearing in the list.

Some important hints

Digital Reverbs

- If available, use digital Ins and outs.
- If only analog Ins/Outs are available, use 24bits sound cards for recording.
- Too low signal level worsens SNR.
- Absolutely avoid digital clipping!
- Keep dry/wet ratio at zero! (only wet signals should be recorded)
- Record the complete reverb tail of the chirp.
- Avoid recording clicks from switching gear on or off !
- The chirp length normally shouldn't exceed one minute! (Except when any chirp-remains appear. Details for this: see below.)

Natural rooms

- Light distortion of the speakers is OK for longer chirps.
- Avoid digital clipping of the AD/DA converters!
- The louder the chirps are reproduced, the better the SNR is.
- The longer the chirps are, the better continuous disturbing noises (e.g. traffic noise or hiss) disappear from the result. The SNR of the final impulse response improves significantly in comparison to the SNR of the reverberated chirp if the used chirp is longer than one minute!
- You can also use longer chirps to obtain good results with a lower playback volume. That keeps speakers, ears and nerves much healthier!
- Absolutely avoid impulse-like disturbances while recording, like coughs, moving chairs or falling objects. Otherwise it will result in remains of the chirp in the impulse response.
- The frequency characteristics of the microphone, amplifier and speaker are transferred to the result. So use high-end gear, if possible. The speaker normally is the weakest link.

Problems and solutions

Chirp remains in the impulse response

In some cases there are remains of the chirp in the calculated impulse response. Two typical causes and its solutions follow:

Cause 1: Aliasing

When using older digital reverb units, which are working with a low internal sample rate, aliasing can happen. In a spectrogram visualization of the reverberated chirp you can see the disturbing chirp, which is mirrored in the frequency domain when adjusting the visualization's range to - 90 dB.

Solution
Experiment with different sample rates or try to use digital Ins/Outs to reduce the aliasing component.
Apart from that the only solution is to use a very long chirp. The remaining chirp in the impulse response should now be running through a very narrow frequency range (e.g. from 12000 to 12100 Hz) and can be removed with a steep band pass. Very suitable for this purpose is the FFT-Filter.

Cause 2: Erroneous impulses from the recording, reverberation or signal transmission
Impulses or impulse-like acoustic noises also result in chirp remains in the calculated impulse response. The procedure reacts very sensible to such disturbances. Impulses with a level of near
-50dB will already result in unwanted disturbances.
Typical sources are:
– Chirp playback or recording glitches caused by buffer or driver problems
– Cracks from starting: The chirp has to be played from the very beginning, otherwise the playback starts with a non-zero value, which always results in a crack.
– Crack by stopping the recording too early. Same thing as above: The last played sample is not zero. After stopping the output is zero which results in a crack.
– Impulse-like disturbances such as coughs, moving chairs, doors or falling objects
'– And last but not least: defective cables and plugs!

Solution
To find the cause of the disturbing impulses, listen carefully to the reverberated chirp! If the cracks are reverberated too, the problem is at the playback side. If they are dry, it is a recording problem. In the spectrogram view of the visualization you can realize impulses as vertical lines.

Performance problems when calculating impulse responses from long chirps (longer than 1 minute)
The calculation of the impulse response from longer reverberated chirps is extremely performance consuming. Also the memory need is high. For chirps with a length of six minutes (44.1 kHz), an amount of about 700 Mbytes is needed.

Solution
Make sure that there is enough memory available. If necessary, increase virtual memory. If virtual memory is used, the calculation time will rise significantly (swapping!).
The calculation of a six minutes chirp on a P4 2.4 GHz with 512 MB RAM will last about ten minutes.
The progress indicator of the calculation behaves non-linear – it will become slower and slower, because the amount of necessary operations increases at the end of calculation.

It is recommended to perform the calculations via batch processing (e.g. overnight).

Limitations of the impulse response procedure when simulating digital reverb units
With the impulse response procedure you can only simulate so-called linear and time-invariant systems.
Effects such as chorus or flanger change are frequency response time-dependent. They behave time-variantly and so it is not possible to simulate them. This applies also to modulated effects (e.g. by an LFO).
Distortion and compressors have a non-linear behavior and cannot be simulated as well.
The reverb presets from multi-effect units often use other effects besides the reverb effect, for instance chorus, in order to "camouflage" the unwanted resonance frequencies, which are typical for digital reverb algorithms. In these cases an exact reproduction of the reverb preset is not possible.
The genuine strength of the procedure is the simulation of natural rooms in a very high quality!

“Play/Rec” Menu

Play Once

This option plays the current project (Wave Project or VIP) one time. Playback will start at the current play cursor position.

Shortcut: *Spacebar*

Play Loop

Plays the current project (Wave Project or VIP) as a continuous loop. Playback will start at the current play cursor position.

Shortcut: *Spacebar*

Play With Preload

If you require playback to start instantly, choose this option to preload the audio buffers, then click OK to start playback. This function is useful when you want to synchronize something ‘manually’ and an accurate playstart is required

Shortcut: *Shift + Spacebar*

Play In Range / Loop

This option will start playback at the beginning of the audio tracks, or the current song position, and continues until the end of a selected Range is reached. Once the end of the Range is reached, playback then continues from the beginning of the Range. This is also a quick way of testing sample loops.

Shortcut: *Shift + P*

Play only selected Objects

Only selected Objects will be played. All unselected Objects will be muted temporarily. If the play cursor is not located on the first selected Object’s time position, it will be moved to this position.

Shortcut: *Ctrl + Space*

Play cut → Play to Cut start (InPoint)

The range up to the beginning of the marked Range will be played. The duration of this cut can be specified using Options > Program Preferences > Set Preroll Time or in the pre-settings in the Crossfade Editor. The settings are consistent.

Shortcut: *F5*

Play Cut → Play from Cut start (InPoint)

When simulating a cut using a marked Range, choose this option to play back a short segment of audio, starting from the beginning of the marked Range (Cut Start). The duration of the playback is defined by Options-> Program Preferences->Set Preroll Time or in the pre-settings in the Crossfade Editor. The settings are consistent.

Shortcut: *F6*

Play Cut → Play to Cut end (OutPoint)

Choose this option to play back a short segment of audio that stops at the end of the marked Range (Cut End). The duration of the playback is defined by Options-> Program Preferences->Set Preroll Time or in the pre-settings in the Crossfade Editor. The settings are consistent.

Shortcut: *F7*

Play Cut → Play from Cut end (OutPoint)

Choose this option to play back a short segment of audio that starts at the end of the marked Range (Cut End). The duration of the playback is defined by Options-> Program Preferences->Set Preroll Time or in the pre-settings in the Crossfade Editor. The settings are consistent.

Shortcut: *F8*

Play Cut → Play over Cut/Crossfade

VIP: A cut is simulated. Playback will start slightly before the marked Range, then will skip the marked Range (Cut), and end just after the Range. The duration of the playback is defined by Options-> Program Preferences->Set Preroll Time or in the pre-settings in the Crossfade Editor. The settings are consistent.

Shortcut: *F4*

Stop

Choose this option to stop playback (or recording). The playback cursor will return to the point at which playback commenced. This point can be set using the playback parameters (P).

Shortcut: *Space*

Stop and Go to Current Position

Choose this option to stop playback (or recording), and have the cursor remain at the point the stop command was issued.

Shortcut: *"," or "decimal" (Num Pad)*

Restart Play

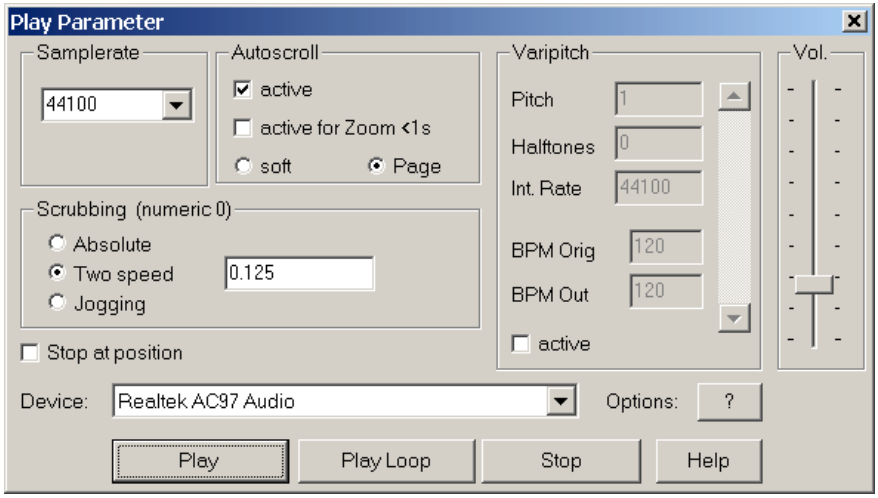
Choose this option to restart playback from the original play cursor position.

Tip: You can select this option while audio is playing.

Shortcut: *Backspace*

Playback Options

Choose this option to launch the play parameter window.



Sample Rate

The sample rate at which a project is played back can be changed here, provided that the soundcard supports this rate. Use this if you want to play back a project that has a sample rate that your sound card doesn't support. You can change the playback sample rate here and then compensate the tempo change with the varipitch function.

Options

Opens a properties screen, which displays which playback features (sample rate, resolution, hardware pitching ...) the selected driver supports. There is also a button to open the Windows multimedia mixer in playback mode (if your soundcard provides one) to quickly adjust the output level.

Device

This allows you to define the device used for playback of Wave Projects and the stereo master device for VIPs. Devices that show up in this list are set up in the Options Menu -> System/Audio settings

Autoscroll

The 'Autoscroll' section activates the autoscroll feature, which is especially useful when working with long disk files. The Wave Project window will follow (scroll) the play cursor during playback.
When working in '2' or '3' section display mode, the autoscroll feature causes the individual sections to follow the play cursor. If you have zoomed into any of the sections, the play cursor will move through the section faster, resulting in additional screen redraws.
There are two autoscrolling options:

Soft: The 'Soft' option smoothly scrolls the entire wave form, with the play cursor remaining centered in the display. This mode requires a fast graphics card, as the entire screen is scrolled between the marker steps.

Page: The 'Page' option performs a page by page scrolling.

Please note that autoscrolling processing power requirements are based on your CPU, graphics card and display resolution. Given these requirements, you may encounter small interruptions to audio playback. Should this occur, simply disable the autoscroll feature, or increase the buffer size (select 'Setup > System').

You can switch on/off auto scroll mode via the "AutoScroll" ("Scroll" key) menu option. The "AutoScroll Soft" (Shift + "Scroll" key) menu option toggles between "page" and "soft" mode.

Scrubbing

You can "scrub" audio by pressing + holding the 0 key, on the numeric keypad (with Numlock active), and moving the mouse left<->right. Playback starts at a very low speed, with the mouse position – relative to the play cursor – responsible for controlling the scrubbing speed.
There are three scrubbing modes available via this dialog:

Jogging: The distance between the play cursor and the mouse determines the playback speed.

Two Speed: Scrubbing takes place at two speeds. Normal playback speed occurs when you drag the mouse quickly over the project. When you drag it slowly, the alternate (second) speed is used. You can specify a playback speed factor (for the second speed) in the numeric field. The default is 0.125, i.e. 1/8th of normal speed.

Absolute: The position of the mouse in the window determines the playback speed. At the left window border, playback speed is 200% backward, at the right border – 200 % forward, in the middle of the window, the speed is 0.
Realtime resampling is performed when changing the playback rate. This occurs without changing the sample rate of the sound card. For best performance, use small play buffer sizes and a fast processor.

Note: When working with smaller buffer sizes (4000, 2000 samples), scrolling becomes 'softer'. Please verify the performance of your computer system at smaller buffer sizes to avoid any interruptions to playback. You may need to experiment with several of these small settings, to ensure error-free playback.

Varipitch

Samplitude supports smooth changes to pitch during playback, even in multi-track projects.
Activate the varispeed mode (with the 'active' button). This will allow you to adjust the playback speed in the following ways:

Vertical slider: Changes the playback speed from -200% to +200%

Pitch Factor: Lets you manually specify a pitch factor

Halftones: Lets you specify a value for halftones. Playback will be transposed by the user defined number of halftones.

Internal Rate: This setting defines the sample rate used for the varipitch calculation. If you wish to play a 48 kHz WAV file, but your sound card only supports rates up to 44.1 KHz, simply set the internal rate to "48", and activate varispeed. The audible results are identical to those you would hear if you were playing back via 48 KHz hardware.
This function is also very useful for digital playback to DAT, at 44.1 KHz, and vice-versa.

BPM: Use these two boxes to enter the original BPM value of your material, and a destination BPM value. The destination BPM value is then achieved via the varipitch function.

Shortcut: *P*

Playback Mode → Loop Mode

Playback is performed in Loop Mode, which means the playback continues from the beginning of a range after reaching the end of a Range or the project. This menu item is mainly present to map a keyboard shortcut on it. You can also switch the loop mode on/off via the LOOP button on the transport control.

Playback Mode → Change Playback Direction

Choose this option to change the direction of playback (i.e. reverse).

Playback Mode → O-Tone Mode

The O-Tone Mode is a special live playback mode that fulfils the requirements of radio editors, but is also useful in other situations.

When o-tone mode is activated, the Objects on the first track of the VIP will be played back sequentially. When playback is started, an area comprised (precisely) of the first track object will be marked and played back.
When playback stops, the next Object will be marked. When playback is restarted, the next Object will be played back as well. Following playback of the last Object, the first Object will be marked again.
This facility allows separate audio Objects positioned on the Track (and separated by pauses) to be played back one after another, at the push of a button.
You can also play back multi-track projects in O-Tone Mode - e.g. surround sound for theaters, or other live projects. Only Objects on the first track will be used for the start/stop markers, so please ensure that the desired Objects are placed on the first track.
O-Tone Mode can also be used when the Object Manager is activated. In this scenario, the Object list can be used for quick selection of the next Object you wish to play.
This is perfect for mixing live speech with pre-recorded material (the ‘original tone’).

Playback Mode → Auto scroll

Switch Autoscroll on/off. For more information on auto scroll, please read the “Autoscroll” section, in the playback parameters chapter.

Shortcut: *Scroll*

Playback Mode → Autoscroll soft

Switch Auto scroll between “page” and “soft” modes. For more information on auto scroll, please read the “Autoscroll” section in the playback parameters chapter.

Shortcut: *Shift + Scroll*

Scrubbing active

Switch scrubbing on/off. For more information on scrubbing, please read the “Scrubbing” section, in the Playback Parameters chapter.

Shortcut: *Alt + Shift + Arrow down*

Jog/Two Speed/Shuttle

Switch between the three scrubbing modes. These menu items are mainly for mapping keyboard shortcuts. For more information on scrubbing, please read the “Scrubbing” section, in the Playback Parameters chapter.

Scrub left /right

To scrub left or right with the keyboard.

Shortcut: *Scrub left* *Alt + Shift + Arrow left*

Shortcut: *Scrub right* *Alt + Shift + Arrow right*

Record

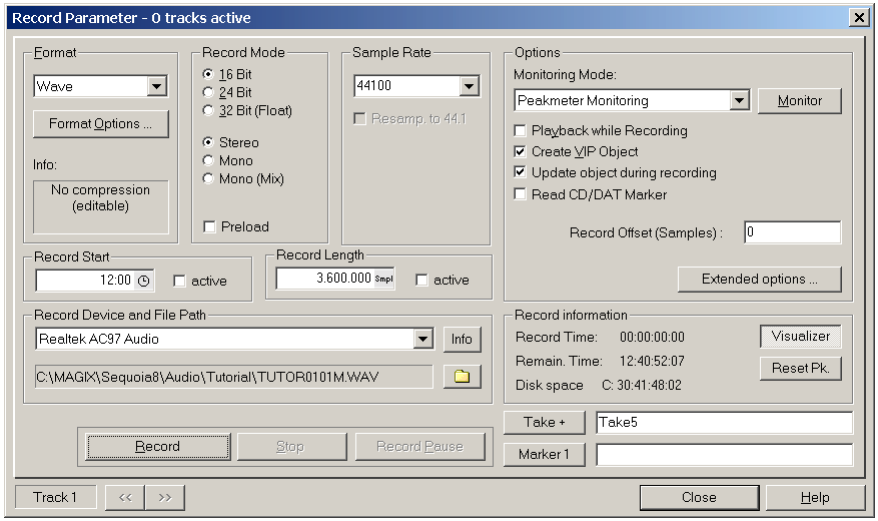
This menu item starts recording, immediately.

A note about multi-track recording: Samplitude can simultaneously record as many files (tracks) as your sound card/s and system can deliver.
Before starting recording, you need to select the appropriate input device for each track. This can be done via the Track Info dialog, or by right-clicking on the 'R' button in the VIP track.
Select the tracks you wish to record to with a mouse click.
Recording then occurs as per stereo/mono recording.

Shortcut: *R*

Record Options

This menu option opens the Record window. All recording settings can be made here. Once set, you don't need to reopen the dialog, and can start recording by pressing the Record button on the transport control, or by clicking the toolbar icon.



Recording and playing are two independent threads that can take place independently. This allows you to close the recording dialog, after recording has commenced. When live updating of the recorded Object is enabled, you will see your recorded audio material appear in the VIP, in real-time. You can freely play and edit your project, while recording is taking place!
Attention: As you can start/stop the playback of the VIP, independent of recording in the background, you cannot stop recording by pressing the space bar! To do so, you must click the Record button/icon again, or re-open the Record dialog and click "Stop"!

Shortcut: *Shift +R*

The settings for most Recording tasks can be made here.

Record Format
You can specify the format to record in. This allows you to record directly in mp3 or AIFF format, for example. The Format options dialog allows you to select the data format you wish to save in - e.g. bit rates or sampling resolution. Some Internet formats (e.g. Real audio) can only be recorded as is - i.e. they are "non-editable". Normally you won't use these non-editable formats in projects, and will record audio in the wave format, and export it to the compressed format, after the fact. By recording in a compressed format (e.g. ".wma"), you avoid the extra conversion step. It is not possible to record in compressed audio formats during multi-track recording.

Record Mode
The radio buttons in this section are used to select either stereo or mono, whether you want to record the audio into a RAM Wave Project or as a Wave Project, and whether you want to record in 32-bit float, 24-bit or 16-bit format.

16 Bit: Select this option to record in 16-bit format.

24 Bit: Select this option to record in 24-bit format.

Float: Select this option to record in 32-bit Floating Point format (IEEE).

Stereo: Click on (enable) this option to record from a stereo input device into a stereo Wave Project.

Mono: Click on (enable) this option to record into a mono Wave Project. If you select this option, the audio recording device selector changes to mono devices, actually the left or right channel of a stereo input pair. Make sure you select the right one (Left or Right channel)

Mono (Mixdown): Click on (enable) this option to record from a stereo input device into a mono Wave Project by adding the contents of both channels and applying a -6 dB attenuation.

PRELoad: Click on (enable) this option to load all buffers before recording begins. This enables you to instantly start recording when the *Record* button is clicked. If disabled, there will be a short pause (depending on the size of the Buffers) before recording actually begins.

Sample Rate

This option allows you to select the sample rate of the recording. The most common sample rates are selectable as (radio button) presets. You can also type the desired sample rate directly into the box.

Important note: Your audio card MUST support the selected sample rate or recording will not work.

Resamp. To 44.1: Select this option, and Samplitude will perform a real-time Sample Rate Conversion to 44.1 kHz. – i.e. if you have a DAT tape that was recorded at 48 kHz, and plan to master its contents to a Red Book audio CD, this option can be used to convert the sample rate to 44.1 kHz while Recording the DAT into Samplitude.

Note: Real-time resampling is not always the best option.
It should also be noted that higher quality Sample Rate Conversion algorithms are available in Samplitude. These are performed off-line, after the file is recorded (the highest quality algorithms are too taxing for real-time processing during recording).

Record Start Time/Length

This box enables selection of a specific duration for recording in hours, minutes, seconds and milliseconds. The recording take will automatically disengage when the specified duration is reached. To enable this function, check (tick) the *active* box, and enter the desired take length in the numeric box. Note that *Remain. Time* is also activated (see above)

Record Device and File path

Input Device: Select the desired input device in this field. Multiple sound card inputs will be displayed here. By clicking on the "Info" button you can check on the record capabilities of the selected sound card. It shows you information on the sound card driver and it's capabilities.{bmc RightW.bmp}Record Device Dialog. You can customize the device names or disable devices you never in the System/Audio dialog, Record Tab.

Note: If you do not see your audio card listed in the drop-down list, go to the System/Audio Settings (key y) and check, if the right driver Model is used and the device is enabled at all. It also can help to press the Reset *button there!*

File Path: This dialog allows the user to determine the name and hard disk destination of the Wave Project. The path to the left of the button indicates the destination of the audio file to be recorded. If you click on the yellow folder button, you may enter a new name or location for the audio file. This is an easy way to redefine a target directory or hard disk as the destination for the new audio file. If you create a new name, a new Wave Project with that name will be initialized. Usually, you don't have to change this because you can define auto-naming rules in the Track Information dialog.

Info (Recording Device Capabilities Dialog)

This dialog allows you to determine the recording capabilities of the selected recording device.

Sample Rate/Resolution: Boxes for supported sample rates/resolutions are checked. For WDM drivers, the reported capabilities depend on whether or not WDM compatibility mode is set (in the System options - Shortcut: y).

Zero Latency Monitoring (ZLM): Pressing "test ZLM support" will indicate whether or not your device supports Zero Latency Monitoring. You can activate Zero Latency Monitoring in the Recording Options section of the Recording Parameters dialog.

Device Info: Indicates the system name of the recording device. Also allows you to change the name (used by Samplitude) of the recording device, listed in the track information/mixer panels. ("System/Global audio options" dialog - Shortcut: y). Driver version and information is also displayed.

Open Windows Mixer: Opens the recording mode of the Windows Multimedia mixer panel (if your soundcard provides one), allowing you to quickly adjust the input level.

Monitoring Mode / Monitor

Most sound cards allow you to preview the audio signal you are about to record, and (depending on the features of the sound card) you will be able to monitor the recorded signal during recording. For more Information about monitoring, read the appropriate chapter below (Menu Play/Rec ->Monitoring)

Playback while Recording

Check (tick) this box to record the Wave Project while listening to previously recorded tracks.

Record Offset

If you have a record offset enabled, Samplitude will automatically compensate for the offset. The numeric box, record offset (samples), allows you to specify an offset which is calculated in samples.
In some rare cases and on some systems it is necessary to compensate for any delays caused by processing speed and other factors, such as your sound card's ability to switch between Playback and Recording mode. Delays can occur between the recording and playback of audio when the *Playback while Recording* feature is turned on. This option allows adjustments if this occurs. Good values to start with are in the 20-2000 sample range – experiment to see which setting best suits your device/system.

Create Vip Object

Check (tick) this option to automatically create an Object (in the VIP), after Recording a new Wave Project. If you check this box, a new Object is created

for every recording take and inserted at the cursor location/in the Track(s) enabled for the recording. If the box is unchecked, Samplitude will not create an Object.

Update Object while recording:
Enable real-time updating of the recorded Object in the VIP. If disabled, the object will be created after the recording stops.

Read CD/DAT Marker
If your audio device supports the input of CD or DAT markers, Samplitude will note the position of these markers in the recorded project. Check (tick) this box, to enable the capturing of CD/DAT markers.

Record Options → Extended Options

Several additional options can be reached by opening the "Extended record options":

"Show small record window during recording/punch in recording"
You can display a small non-modal recording window with the main recording controls while recording. Deactivate the option if you don't want this (V6 behavior).

"Each take in a new file"
You can decide if you want Samplitude to create a new file for each take. These auto-created Wave Projects are named with the original Wave Project's name and a suffix "_Txxx", where XXX is the take name.

Set cursor to record end position
After finishing a recording, the play cursor is replaced to the recording's end position. So the next recording will start at this position. This simulates recording on a tape machine.

Record button in Record Options can stop recording
Disable this option when you experience problems by accidentally clicking the record button twice.

Show confirmation dialog (OK, delete) after record
You can disable the confirmation dialog (Use the recording or not?) by unchecking the box if it should annoy you.

Resampling Options
Quick access to the resampling quality settings, might be necessary when performing real-time resampling, e.g. recording from DAT into a 44.1kHz VIP.

Visualizer
Click on this button to open the Visualization window. (see Window Menu chapter for a detailed description)
To minimize system requirements, you may wish to disable the Monitor function, if it is not crucial to the recording. This may be necessary if you are pushing the limits of your system.
During recording, the meters will be slightly delayed in their reaction, due to the priority level change. The most important task is to keep your recording(s) error free. As such, the updating of VU meters is a lower priority task. This is to avoid overloading the computer. The levels are still shown at their correct values. If you use lower buffer settings, the meters are updated at shorter intervals (i.e. they are faster).

Tip: Adjust the Input signal's level so that the loudest peak doesn't quite reach 0dB. The Input level of the signal must be adjusted at the source or with a mixer. Some audio cards also allow you to adjust the input signal.

As with all digital audio, any levels above 0dB are clipped, resulting in ugly distortions. If a red line is displayed at the up, we suggest that you lower the input signal level. If this happens during recording, you will probably need to re-record the take. If the take has been completed, however, we suggest listening to the file before deciding to discard it – the clipped section may not be significantly damaged.

Note: Samplitude offers a de-clipping feature in the Effects menu. This allows for audio repair in cases where re-recording is not possible. This should only be used as a last resort – it is always better to avoid clipping during recording, if possible!

Reset Peak
Click on this button to reset the LED Peak-Meters to the off position.

Record Information
This display contains 3 numerical counters which show the following data for recording:

Record Time: This counter indicates the length of the current recording in hours, minutes, seconds, and milliseconds.

Remain. Time: This counter shows the remaining space on the hard disc in terms of time units. This counter is updated in real-time during recording.

Disk Space: This counter displays the remaining space available for recording on the specified partition in hours, minutes, seconds and milliseconds.

Take
A "take name" (Object name) can be set for the Object, prior to starting recording. This can be done by entering a name in the text box to the immediate left of the "Take " button. Each take is named with this name and an ascending take number. When the Take button to the right of the text box is clicked, the take number is increased by one.

Marker
You can drop markers into the project during recording. The current cursor position is used to insert markers at positions you may wish to select later, for clean-ups, or similar tasks.
Each marker can be named (if you have time!) by entering a name into the adjacent text box.

Record button
Start recording by activating this button.

Record Pause
Pauses recording until pressed again.

Stop
Stops the recording process. Samplitude will ask you whether you wish to keep or delete the recording.
Should the computer become overloaded due to data swapping or hard disk access issues, for example, simply interrupt the recording with a right-click or the Spacebar.

Track
You can switch between the tracks and so adjust the recording parameters for all tracks in one go.

Close

Close the recording options dialog. You can open and close this dialog without affecting an actual recording!

Record Mode/Punch In → Record without Playbck / Playback while Recording

Switch between recording with or without playback of the other tracks (not armed for recording).
This menu item is mainly intended for mapping a keyboard shortcut on it. You can also switch the record mode via the Audio menu on the Transport control.

Record Mode/Punch In→ Punch In Mode

This option activates the Punch mode. Punching In and Out allows real-time or automatic commencement and suspension of recording at specific points in the timeline.
Once the Punch mode is active, Punch-In record can be used to record audio segments without stopping playback.
The ability to predetermine and automate this process is very useful for musicians or composers who work alone. In addition, Punch In and Out points can be determined at a single sample level, allowing accurate positioning of entry and exit points.
The Punch In mode toolbar (shown below) is very useful when working in Punch In mode.
Punch In mode requires one or more tracks to be armed for recording. The tracks need to be configured in the same way as for multi-track recording – i.e. each track requires a specific device selection. Right-click on the 'R' button of each track to make your input device selection(s).
Punch recording can be done in two ways:

Punch-In/Out 'On-The-Fly'

This allows you to start the recording (punch in) at any time during playback. Once recording commences, it can be stopped at any time (punch out) with playback continuing.
This is very similar to tape based multi-track recorders, which allow you to correct mistakes made during previous recording passes by overwriting segments of the tape.
Activate Punch-in mode. Simply start playback with the Spacebar. Punch-in recording can be started by clicking on the punch record button on the toolbar. The same button punches out of record mode with playback continuing. To stop playback, press the space bar.
(The same function is obtained by clicking the dedicated Punch-In Record button, which is obsolete now but left for convenience)

Punch-In/Out with Markers

The second method involves punching in/out of Record mode through use of special markers. This is a more automated way of dealing with punch-in/out

recording, and is very useful for musicians or composers who work alone.
By predetermining In/Out points, this method also eliminates the risks attached to misplaced manual "punches". Punching In and Out with markers allows you to specify (down to single-sample level) the exact points where the recording is to start and stop.
As you are working in a non-linear, non-destructive environment, any mistake or unwanted take can be removed or undone without problems. Any existing material in the track will still be there.
To operate in this mode, it is helpful if a Range (that defines the beginning and end of the recording), is selected. This instructs Samplitude to stop playback at the end of the Range. (If a Range is not selected, you will have to manually stop playback, after the recording has 'punched out')
The *Set Punch-In marker* and *Set Punch-Out marker* functions are used to instruct Samplitude where to punch in/out.
Position the play cursor appropriately (prior to the punch-in point), and start playback/recording with the Punch Record button, found in the Punch Toolbar.
Note: When working with punch in/outs, it is useful to have the Auto Crossfade mode activated (by clicking on the corresponding button in the Samplitude toolbar). The Auto Crossfade will create smooth transitions automatically between takes.

Deleting or repositioning punch in/out markers

The In/Out markers can be relocated by repositioning the cursor, and clicking on the appropriate Toolbar button. To delete punch in/out markers, click on the *Remove Punch In / Out Marker* Toolbar button.

Looping Punch In recording

Samplitude can also perform looped punch-in recording. Simply select a Range over the desired punch region. The Range is played back until you stop playback with the space bar.
Each time the program loops through the Range, new takes are recorded at the punch markers. Make use of the Take Manager to find the best take after recording has completed.

Record Mode/Punch In → Set Punch In/Out Marker, Remove Punch In/Out Marker

The In/Out markers can be relocated by repositioning the cursor and clicking on the appropriate toolbar button. To delete punch in/out markers, click on the *remove Punch In / Out Marker* toolbar button.
This menu item is mainly present to map a keyboard shortcut on it. You can set the Punch Markers also via the In and Out Marker buttons on the Transport Control.

Record Mode/Punch In → Set Punch Additional In/Out Marker, Remove Additional In/Out Marker

You can define several Punch/Out Ranges in the project, so that you do several Punch In/Out recordings in one go. This menu item is mainly present to map a keyboard shortcut on it. You can set additional Punch Markers by pressing

ALT and clicking the In and Out marker buttons on the Transport Control.
Or drop additional punch markers by moving the play cursor to the desired position and click on the PunchIn/PunchOut key (Transport Console) whilst pressing the Alt key.

Record Mode/Punch In → Live Input Mode

The Live Input Mode is another very powerful feature of Samplitude.
It allows you to mix live inputs from the sound card channels via the real-time mixer.
This turns Samplitude into a full-scale digital real-time mixer with unlimited channels, internal 32-bit processing, external 24-bit I/O (requires compatible audio cards) and effects, including third-party plug-ins!

The Basics

The Live Input Mode is somewhat similar to recording with multiple sound cards inputs. All channels/tracks that are to be used for live mixing need to be armed for recording – i.e. each track must be configured with the correct recording device and the track arming buttons need to be active.
Once playback of the VIP is started, live signals on the inputs of the audio device(s) are routed through the Mixer. Mixer window controls are used to create a mix. VIP Objects are also played back while the live inputs are processed.

Latency

Audio processing in Samplitude obviously takes time. This may lead to small delays at the outputs when heard alongside the incoming signal. This delay (latency) is very much dependent on the VIP buffer size, and can be reduced by using smaller VIP buffer settings.
You should use settings that enable the real-time processing of audio without introducing errors, or audio artifacts. A setting of 4 buffers, at a buffer size of 8000 stereo samples, creates a delay of 0.7 seconds. A reduction in buffer size (to 2000 samples) results in a delay of only 0.2 seconds.
Experiment with the buffer settings to find the optimal settings for your system. In some systems, the latency may make it impossible to use the Live Input Mode as an analog mixing console replacement.

Applications

The Live Input Mode may be used for the following:

Digital Mixer for ADATs (or other multi-track tape machines).

During Playback of an ADAT tape, 8 Tracks are transferred via a digital audio card and processed by the Samplitude Live Mixer. As the material is transferred digitally, there is no quality loss, and the latency is inconsequential, as the ADAT tape does not have any "live" relevance.

Effects and Mix Down during Recording

Use the "Mix to File" function in the Mixer to create a new .WAV file from the master signal. This allows Samplitude to record an audio file via the Mixer. Don't forget to rename the newly created audio file after recording or you will overwrite the contents of the previous take, during the next "Mix to File" process.

Samplitude as Live Effect Processor

If you are looking for processes that are not offered by effect units in your rack, you can make use of some of Samplitude's integrated effects. A live signal can be processed with the FFT Filter, the Multi-Band Dynamics processor, and the Stereo Enhancer. You may also use the many plug-ins available.
New, more flexible input signal monitoring is now available through ASIO driver support.

Input monitoring on/off

Activates/deactivates input monitoring (depending upon the "REC M" button on the transport console). Monitoring renders the input signal of the tracks activated for recording (REC button in the VIP is lit) according to the selected type of monitoring for the output signal. The LED peak meters in the VIP and mixer react to the input signal.

Tape player monitoring

The preset method of monitoring as on a tape player and well-known from earlier Samplitude versions:
When STOP is shown, the input signal is rendered, PLAY renders the track contents, RECORD renders the input signal except with punch-in recordings (which renders that which is outside of the track punch markers and within the input. If the option is deactivated, it is left to the user to choose whether the input signal is shown. Use the "Force monitoring" button (small loudspeaker symbol beside/above the peak meter in the VIP/Mixer).

Hardware Monitoring / Software Monitoring / Software FX Monitoring

The input signal is only given when the monitoring type has been selected. Nevertheless, the peak meter continues to function. When the monitoring type has been selected, audio monitoring is activated.

Hardware Monitoring

With MME drivers, this is the only possible type of monitoring within Samplitude. In addition, the audio hardware must support ZLM (Zero Latency Monitoring). You may of course offset a monitor signal in the Windows Multimedia mixer externally from Samplitude. However, it is incompatible with the punch-in functions.

Software Monitoring / Software FX Monitoring

For cards with ASIO drivers, it is also possible to use software monitoring if the card does not support hardware monitoring (ASIO Direct Monitoring).
Software FX monitoring also calculates the track effects into the monitoring signal with the exception of all high latency-afflicted FX Inserts such as FFT filters, Dehisser or Vocoder. The amp simulator, however, is applicable. Samplitude is therefore also usable as an effects device with which one can play live directly.
If you need all of the mixer effects, use the conventional Live Input mode, which functions without ASIO, but is afflicted with higher latencies.
In addition, software FX monitoring must be active before VST Instruments can be played live via the MIDI IN port.

"MIDI" Menu

New MIDI Object

A new MIDI Object is created on the selected track. After choosing the command, you can select a prefabricated MIDI template (from the subdirectory "Templates" in the Samplitude directory) from a small pop-up menu. It deals with normal standard MIDI files which can be copied into this directory or directly exported from Samplitude as a template. If only one file is in the template listing, no menu appears – the Object is created immediately.

MIDI Editor

This menu point opens the MIDI Editor. Here you can view and edit the content of the MIDI Object selected in the VIP.
If no MIDI Object is selected, you will be asked if you would like to add one. Confirm this question with "Yes" to add a MIDI Object in the actual track at the current cursor position or at the beginning of an area.

For more information on the MIDI Editor refer to the chapter "Working with the MIDI Editor".

Object Editor

Please refer to the corresponding section of the "Menu Object" chapter.

Key: *Ctrl+O*

Glue MIDI Objects

This command lets you glue together two or more subsequent MIDI Objects of a track. As with the audio version of this command, (Glue Objects in the "Objects" menu), all Object effects of the following Objects are lost.
This way, a looped Object can be changed to a new MIDI Object. For example, you can create "real" MIDI data from a looped drum kit pattern for working on further details.

Trim MIDI Objects

If a MIDI Object only refers to a section of a MIDI file or recording, this is represented in the MIDI Editor in grey; unused areas are still available in the MIDI Object and are, for example, available for subsequent length changes to the Object.
This command removes non-used MIDI data to increase the overview within the project.

MIDI Bouncing

All selected MIDI Objects are bounced into a new MIDI Object including all Object and track effects.
You can choose between "whole Object" and "Sel. In Range" (selection/Range bounce).

Global MIDI Bounce: All MIDI data are mixed together. Track and Object effects are accounted for. A new MIDI Object is created in the selected track. The bounced Objects are removed.

Range/Selection Bounce: All selected Objects in the current Range are bounced. Track and Object effects are accounted for. A new MIDI Object is created in the selected track. The bounced Objects are not removed.

Track Information

Refer to the corresponding section of the chapter "Menu Track".

Key: *Alt+i*

Track MIDI Record

The MIDI recording mode is activated for the selected track.

MIDI Controllers

Refer to the chapter "MIDI".

Key: *Ctrl+Alt+A*

VST-Instrument Editor

Refer to the corresponding section of the chapter "Menu Track".

Tempo/Beat markers

Read the chapter "Tempo and Bar Changes" in the "MIDI" chapter.

Place new tempo marker

Read the chapter "Tempo and Bar Changes" in the "MIDI" chapter.

Place new beat marker

Read the chapter "Tempo and Bar Changes" in the "MIDI" chapter.

Place new beat position marker

Read the chapter "Tempo and Bar Changes" in the "MIDI" chapter.

Ignore all Tempo markers, use project tempo only

Here you can temporarily bypass all tempo marker settings. For all tempo-relevant displays (snap and grid) only the project tempo is used.

Metronome active

Switches the Audio/MIDI metronome on/off. You can also click on the CLICK button on the transport control.

Metronome settings

Refer to the corresponding section of the chapter "Menu Options".

MIDI settings

Refer to the corresponding section of the chapter "Menu Options".

MIDI Record Modes

The MIDI Record modes define how the newly recorded MIDI data is inserted into the VIP if there are already MIDI Objects at the recording position.

Normal

The record mode corresponds with the audio recording, i.e. a new MIDI Object is created on top of the existing Object at every recording. The old Objects are retained. This way you can record several takes of a passage and compare them afterwards.

Overdub

The MIDI data is recorded in the existing Object; the new data is mixed with the existing data. If a recording extends over several MIDI Objects, then these are combined into one.

Multi-Overdub

In Multi-Overdub the new data is also mixed into the existing Objects; however, individual Objects remain.

Replace

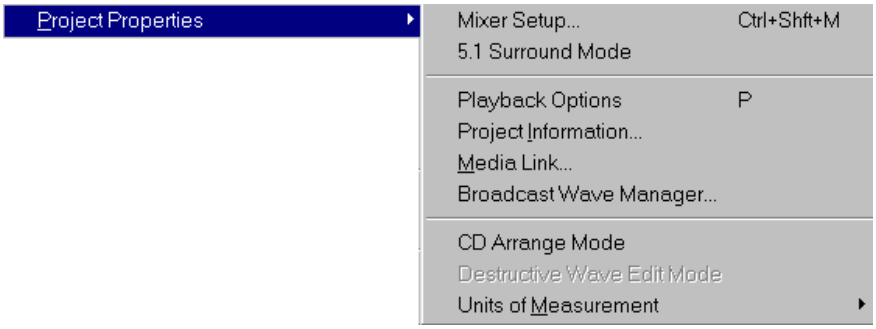
The MIDI data of the existing Objects is replaced. When recording over several Objects, these are combined into one.
In contrast to audio recording, MIDI Objects already on the recording track are played back in "Playback while recording" mode. This means that you can still choose one of the record modes after completing successful recording.
Note: The MIDI Record Mode is also visible in the expanded transport control and can be selected here.

MIDI Panic – Terminate all notes

This command sends a note off command for all 128 notes on all 16 channels, excluding all of the MIDI devices deactivated in the MIDI options. In addition, the sustain (controller 64) is deactivated and the pitch wheel and the modulation are set to 0. An all-notes-off command is also sent to all of the VSTis used in the project.
If MIDI tracks or Objects are available in the project, the same function is accessed by clicking the "Stop" button in the transport control or tool bar (if the project is in stop state).

“Options” Menu

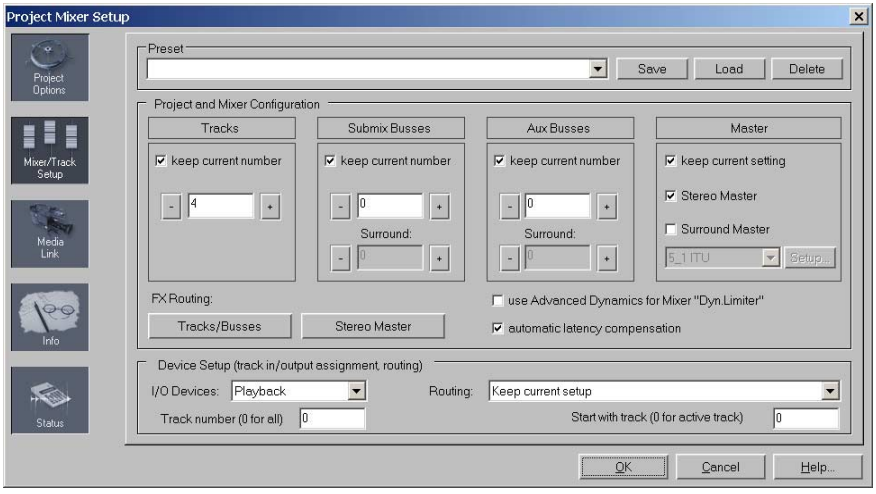
Project Properties



This item provides easy access to various modes, parameters and options.

Project Properties → Mixer Setup

Opens the Mixer Setup dialog, where you can adjust the general settings for the mixer’s routing design. Please refer to the “Mixer” chapter for details. This window lets you easily configure, save and load the VIP and mixer settings.



Project and Mixer Configuration:
Use this to set the number of tracks, AUX busses and submix busses. You can also define effects and signal routing for the tracks and the master.
The window of the first track is always opened for effects routing of individual tracks/busses. In order to reset the settings of all tracks you have to click on the corresponding "All Tracks" button in the "Effects Routing" window.
There are two further options aside from the routing buttons:
– Adv. Dynamics for Mixer "Dyn.Limiter": In the mixer master the resource-saving "normal" dynamics can be used as a Dyn./Limiter or as advanced dynamics of higher quality.
– Automatic latency compensation: Activates automatic latency compensation for DirectX and VST plug-ins. This is activated by default.

Signal (output) routing for the master
– Keep current setting: Each track is assigned to an audio device (or a submix bus). The final mix then takes place in external equipment or on the sound card.
– Stereo Master: Normal mixing occurs in Samplitude's stereo master. If your sound card provides several devices, select one in the mixer (below the volume faders) or in the Play parameters (key: "P").
– Surround Master: If this setting is chosen, master channels are provided for 5.1 surround mastering. Read the chapter "Working in 5.1 Surround Mode" for more information.

Keep current number (tracks, AUX and submix busses)
By checking this box you can prevent the current settings from being overwritten when loading a mixer preset. The current status of this checkbox is also saved when saving a preset. Therefore, if you quickly want to add a few tracks, (e.g. 4 AUX busses and 4 submix busses), you only have to load the "4 Busses +4 Auxes" preset.
So, when loading a preset, the respective tracks and busses where the box is not checked are added. In other words: Place a tick before saving where the settings are not to be included in the preset.

Device Setup (track in/output assignment, routing)
In this section, you can conveniently select the in- and output device assignment for several tracks simultaneously.

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 8-12.

Routing

Keep current setup: No routing assignment takes place if a preset is loaded.

Assign all tracks to stereo master: All tracks (incl. AUX and submix busses) are routed to the master. The panorama settings are alternatively set to left and right via L/R panning. Use this if you are recording stereo sources 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 (key: "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.

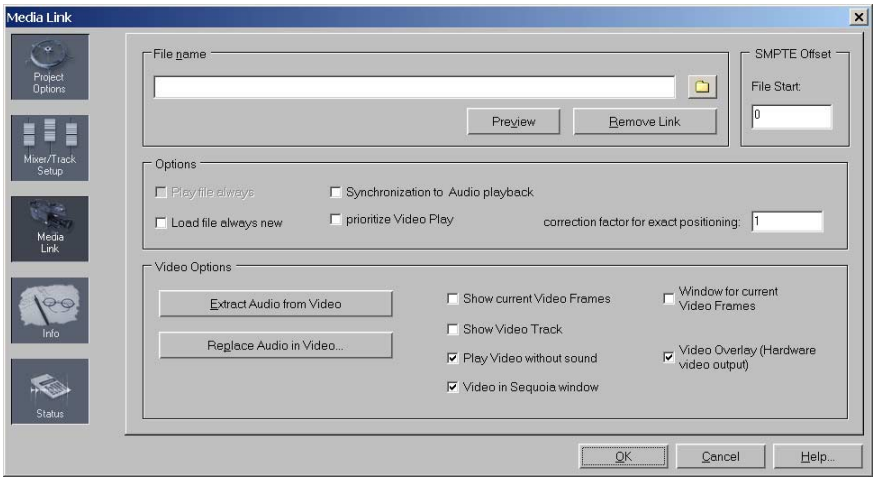
Use saved setup or preset: The settings saved in the preset are used.

Project Properties → Playback Options

This item launches the Play Parameter dialog (shortcut – "P"). Please refer to the *Play/Rec* menu section, for further details.

Project Properties → Media Link

Nearly all other media files can be linked to Samplitude projects. These can be complete MIDI songs (.MID files), video clips (AVI or MPEG files), or other files for which an MCI is installed, practically all formats that the Windows Media Player can play back.



"Linked" means that the linked media files are played back simultaneously with the Samplitude project and that no synchronization occurs. The time position is not verified directly. Samplitude uses the Windows Media Player for playback, which merely starts and stops the selected file. As the media player plays back the AVI or MIDI file, the file must be saved on your local workstation.

This option does not let you synchronize external devices with Samplitude. If you require such synchronization, please refer to the chapter entitled "Synchronization" for all information on conventional synchronization functions via SMPTE, MMC and MTC.
Samplitude is capable of loading MIDI files and provides convenient editing functions. It is no longer required to use the media link together with the MIDI files, even if this is still possible. Playback then is not carried out by Samplitude, but instead by the multimedia subsystem of Windows. The output device is set in the Control Panel. Please note that the Microsoft GS Synthesizer which is frequently used as a standard device for MIDI output prevents the use of ASIO drivers for many sound cards.

Linking Samplitude projects to videos

Media Link makes it possible to add soundtracks to videos with Samplitude in real time without requiring additional synchronization hardware.
In theory it is possible to work with digital videos directly in the original resolution. However, to ensure optimum synchronization and a smooth workflow the following procedure is recommended:

1. Digitalize your source material with "MAGIX Move Edit Pro" video editing software (included in Samplitude Professional) and export a pilot video of small size using a suitable codec which requires low decompression expenditure and only contains key frames if possible. (DiVX or MPEG are not suitable). MAGIX's own MXV format is recommended, as this format is optimized for real-time processing.
2. Link the video to Samplitude and apply the soundtrack.
3. Remove the link again after track bouncing.
4. Replace or store the finished Wave Project in the video using the function "Replace audio in video". You can then save the sound in the original high-resolution movie without requiring recompression (merely the soundtrack needs to be exchanged). More details available from "File" menu -> Exporting video sound...

File Name: Allows you to specify the file name, and source directory of the media file. Clicking on the folder button launches a file selection dialog that allows you to browse your hard disk(s) and specify the media file type. Once you have located the desired media file, simply click on OK, and Samplitude will make the link.

Offset: The Offset allows you to specify a different start time for the media file. The default is a zero position, which means that the media file will start at exactly position 0 (i.e. the beginning of the file). If the offset is varied from the 0 position, playback of the media file starts at the entered position.
As an example, imagine that you're working on some audio for an AVI file, and that this audio section does not occur until 15 minutes into the AVI file. It would be unnecessary to create 15 minutes of empty space at the beginning of the VIP, in order to line up the AVI frames and the audio.

The better alternative is to specify an offset of 15 minutes – i.e. the AVI file will start playback 15 minutes into the video file. All audio in Samplitude, however, will start at position 0.
It makes sense to write down the offset values of each individual project that you are working with.

Play Always: This checkbox allows you to specify whether or not Samplitude should play the media file every time you start playback, or recording. If unchecked, Samplitude will not play back the media file. This is a quick way to disable playback of the media file.

Load Always New: If this option is checked, Samplitude will always load the media file every time you play back, or record to, your VIP or Wave Project. If the box is unchecked, playback of the media file is performed from RAM. If the media file is too large for the buffers, playback is performed from the hard disk.

Synchronization to Audio playback: Synchronizes the new media link (e.g. AVI-Files) with the existing VIP.

Extract Audio from AVI: If the file type is an AVI file with video and audio data, audio streams are extracted from all other information and the audio information itself is put into a Wave Project. This allows you to extract and edit the audio apart from the rest of the contents in the imported file.
You can also edit the audio data from a video file directly (without creating a Wave Project as a copy of the audio data from the video file). Just open an AVI file from the Open Project -> Wave Project... dialog.

Replace Audio in Existing AVI: This calls the Export Video Sound function to replace the audio data in the existing AVI or create a new AVI with new audio.

Play Video Without Sound: This option applies to AVI files only. If the AVI file contains audio, the playback of the AVI file and its audio could interfere with Samplitude's audio playback. For this reason, it is suggested that you disable audio playback for the AVI.

Video in Samplitude Window: This option is only applicable to AVI file playback. Normally, the AVI file is played back in a completely independent window. This means that if you maximize the Samplitude window, you will not see the AVI playback window. You would need to minimize, or scale, the Samplitude window in order to display both windows on screen.
Checking this option, however, allows you to display the AVI window as another Samplitude-related window, which always stays "on top", during playback. This allows you to display the AVI movie any time playback or recording occurs in Samplitude.

Show Video Tracks: This option allows you to display a video track in the VIP window - as individual frames of the AVI file.
In the upper left-hand corner of the VIP window, the current frame is displayed when the play cursor is clicked/positioned at a particular location in the VIP tracks. This allows you to position audio events and other material at exact frame positions.
If you zoom in to the window, the video track is also enlarged, showing the video frames at a lower resolution level. If you display the window at its lowest zoom level, each frame is shown in the video track.

Correction factor for exact positioning: This option allows adjustments to the small delays that can occur on slower computer systems. The linking of media files with VIP or Wave Projects is based on two playback processes: one for the Samplitude window and one for the media file window.
No synchronization of material takes place during the playback process. This means that on less powerful computers, the digital audio tracks and the media file may run 'out of sync'. The FX option allows you to specify a factor, used for internal sample rate conversion.
Here's an explanation of how it works:
As we cannot influence the playback speed of the media file, we need to adjust the playback speed of the digital audio tracks. Imagine that a normal playback speed is represented by a factor of 100. If the digital tracks play ahead of the media file, playback needs to be slowed down.
This would be accomplished by multiplying the regular playback speed of 100 with an FX factor just below 1 (e.g. 0.998). This would result in a playback speed of 99.8, which is slightly slower than the regular speed. This way, we can align the digital tracks with the media file.
On the other hand, if the digital audio tracks lag behind the media file playback, a factor higher than 1 must be entered (e.g. 1.002)
The "rule of thumb" for this factor is to use small value increments. The values should stay around 1. You will need to experiment with this setting, as there is no "one-size-fits-all" rule that can be applied here.
If you don't have any sync problems with the digital audio tracks and the media file, you should stay away from this field!

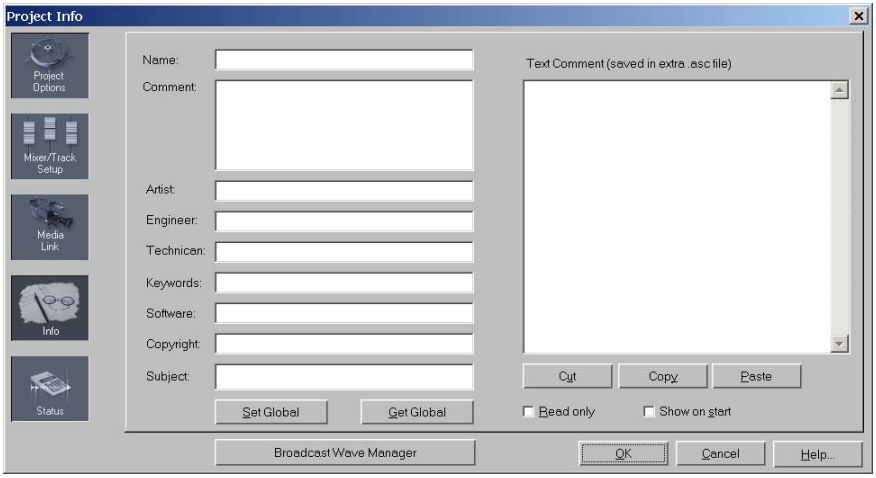
Remove Link: This button will unlink the media file from the Samplitude window. This effectively stops all association with the media file. Use this option if you no longer need to reference the media file.

Preview: The "Test" button allows the current link to be tested. Clicking on the button should commence playback of the media file. If no playback is visible in a Samplitude window, you may need to check the "Video in Samplitude window" option. Other media file types should play back via the Windows multimedia extension, in utilities such as the MIDI file player.

Show Actual Frames: This option shows the actual video frames in the preview windows of the video track.

Project Properties → Project Information

Select this option to display project Information (an "overview") about the currently active project (HD Wave, RAP, or VIP).
This includes creation date, memory size, path and file names.
For Virtual Projects, a list of all included physical files (including linked Wave Projects) is displayed.



The Project Information window displays information in one of the following two ways:

1. Project Information window (HD Wave, RAP)
For Hard Disk and RAM Projects (HD Wave, RAP), the Project Information window will display: path (location on hard disk), file name, number of stored Ranges, number of stored cursors (markers), date the file was created, date the file was last edited, size of the file (in samples), and amount of hard disk space (memory) the file requires. You can also add (type in) the following information: Name, Comment, Subject, Artist, Engineer, Technician, Keywords, Software, and Copyright.

Note: This is a very useful option for users who work with banks of loops, Foley/library sounds, station IDs etc. These "sound libraries" can often appear in multiple VIPs and can also be used by multiple engineers/producers. By setting important files as Read Only, multiple users can work without the risk of altering important files or library files. Additionally, when using the Remove Unused Samples option (Tools > Remove Unused Samples), files marked as Read Only will not be touched.

2. Project Information window (VIP)
For Virtual Projects (VIP), the Project Information window will display: file name, path (location on hard disk), number of stored Ranges, number of stored

cursors (markers), date the file was created, date the file was last edited, number of Objects in the VIP, and a list of ALL Wave Projects used in the VIP.
You can also add (type in) the following information: Name, Comment, Subject, Artist, Engineer, Technician, Keywords, Software, and Copyright. This dialog also allows you to set automatic backup options which run while the Project is open.

Set Globals: Click on this button to save the information you added (typed in) as a global setting.

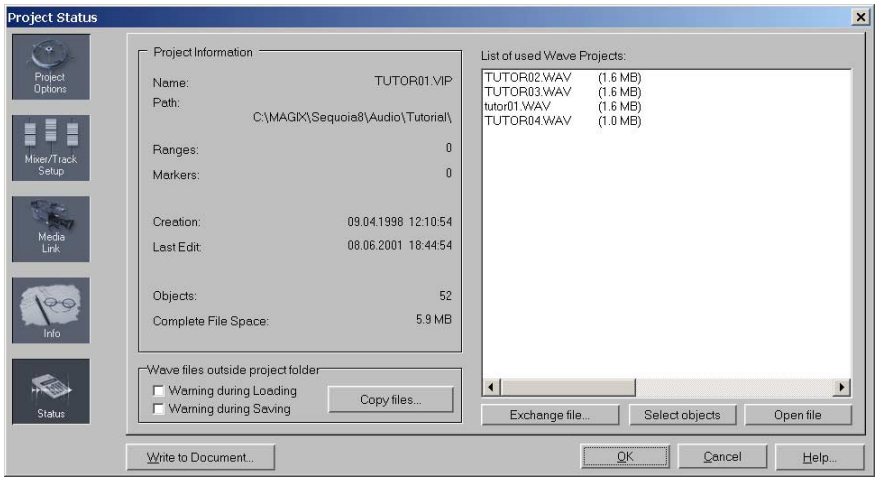
Get Globals: Click on this button to recall the saved information (global setting), and add it to the current active project. This allows you to quickly add the same information to every project you're working on.

Cancel: Click on this button to close the Project Information window, without accepting changes to the settings.

OK: Click on this button to accept the current settings, and close the Project Information window.

Read Only: Click on (i.e. enable) this option to make the HD Wave or RAP a "Read Only" file. You can play the audio file, but cannot alter it in any way.

Project Status



Select this option to display project status (an "overview") about the current active project (HD Wave, RAP, or VIP). This includes creation date, memory size, path and file names.

Project Information

For Virtual Projects, a list of all included physical files (including linked Wave Projects) is displayed. This dialog also shows you the number of markers and

stored Ranges that are used. The information about creation and last edit and how many Objects are contained in the VIP are displayed too.
For Hard Disk- and RAM Projects (HD Wave, RAP), the Project Information window will display: path (location on hard disk), file name, number of stored Ranges, number of stored cursors (markers), date the file was created, date the file was last edited, size of the file (in samples), and amount of hard disk space (memory) the file requires.

Original Time Position (only Wave Projects)

When you record or import some audio material in your VIP, a Wave Project is created, which contains the audio data. Samplitude saves this time position, where the Object was created.

List of used Wave Projects

All used Wave Projects are listed here. You can open the relevant Wave Project window by double-clicking a list entry or choosing Open file.

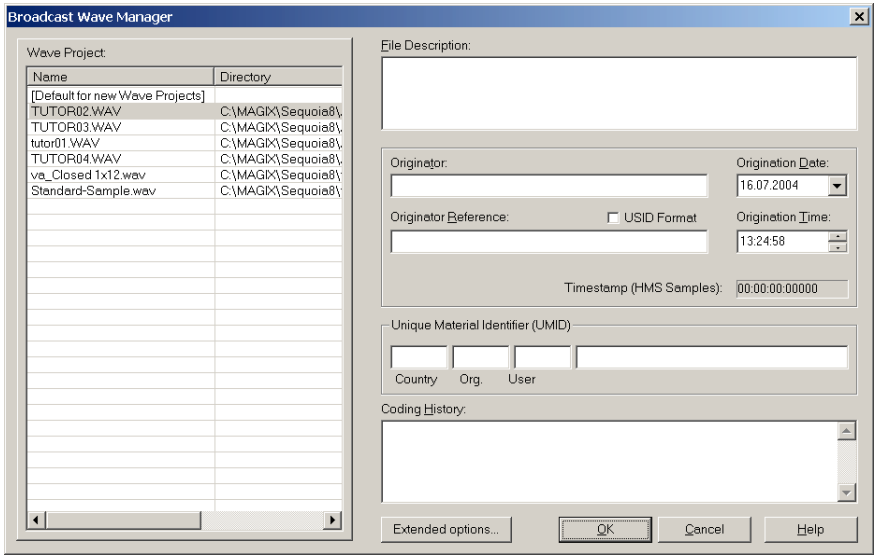
Exchange file: You can exchange a Wave Project in all referring Objects at once here

Select Objects: Selects all Objects referring to the selected Wave Project
You can activate warnings for loading or saving, when the project contains Wave Projects, which reside outside the project directory.

Copy files opens the "Collect project files dialog"

Project Properties → Broadcast Wave Manager

The Broadcast Wave extension lets you save information on an audio file, so-called meta data, in a chunk of the BWF file. This meta data can generally be employed proprietarily. However, we recommend that you observe the guidelines issued by EBU and SMPTE, respectively.



Menu: Options>Project Properties>Broadcast Wave Manager.
By default, you can open this window via the Alt+Shift+B shortcut or as a submenu of the Project Properties.

Wave Project (left half of window)

This displays a list of the data included in the VIP. Select the respective file using the mouse to display individual information. The corresponding BWF information is then displayed in the right half of the window.
The main application of the BW Manager, however, is defining the meta data which is to be included in the Broadcast Wave Extension of the material that is to be recorded.
This meta data 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.

List of the different fields (right half of window)

All values set by the user are saved referring to the VIP and are applied to new audio data.

File Description: Text box that accepts any entry. You can enter a short and a long title here. A maximum of 256 ASCI characters can be entered.

Originator: Contains details on the origin of the file, e.g. the description of the producer. 32 characters max..

Originator Reference: Is determined by the Originator. This can be an internal reference number. Within the scope of the EBU, a recommendation according to EBU Recommendation R99-1999 on how this field should be structured applies here. Activate the "USID Format" option to apply this property to the entry. The entry can then be formatted according to the EBU recommendation (max. 32 characters).

Origination Date: Displays the date on which the file was created. This can also be edited. This makes sense, for example, if material has been stored to the file for the first time, but has been recorded some time ago and that date is to be used as reference. If the entry is edited in the BW Manager, the resulting value is independent of the file properties.

Time: The time at which the file was created is entered here. Like the date, this is automatically created from the file properties but can also be edited due to the same reasons mentioned above.

Timestamp: The timestamp saved in the 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 can provide information on the time on the day of recording.

Unique Material Identifier (UMID): Handling of the UMID is regulated by SMPTE. An appropriate document is available from SMPTE. The code number is SMPTE 300M-2000. We recommend that you observe this guideline as well as the agreements regarding the use of the UMID before using this feature, which are to be agreed for your individual case. Use of UMID is not absolutely necessary for a valid BWF.

Coding History: Besides including details on the file format (A: coding, e.g. PCM; F: Sample rate; W: Bit width; B: Bit rate; this value is only used for non-transparently coded material such as MPEG or MP3; M: channel number;) each entry in this field contains the value T:. This is a comma-free text string, which, for example, allows for the entry of the serial number of the recording analog tape machine, codecs, dither types, AD converters or special signal processing applied to the file such as denoising.
An entry is made when a file is recorded in Samplitude. If this file is then processed again (e.g. the bit width has been reduced or encoded to MPEG), a further entry is added.
Use of the coding history is regulated in the EBU Recommendation R98-1999.

Extended 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).

Reading graphic data and using such data in Samplitude: If this option is activated, the peak chunk included in a BW file is used instead of a .ho file.

Project Properties → CD Arrange Mode

When you activate this menu item, Samplitude re- arranges newly inserted Objects, and inserts pauses between the Objects (conforming to the Red Book Standard).
Please refer to the *CD > CD Arrange Mode* section for more details.

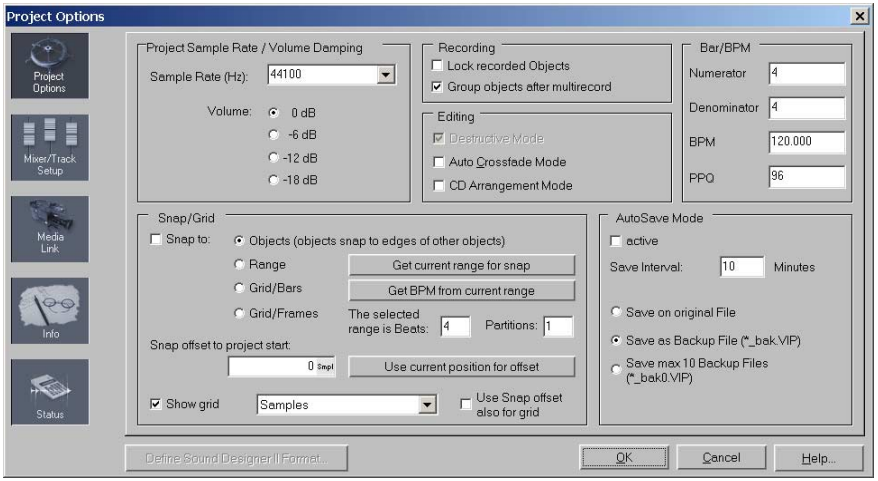
Project Properties → Destructive Wave Editing Mode

This menu point 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.

Project Properties → Units of Measurement

This dialog determines the timing method used for the display/grid.
Please refer to the *View > Units Of Measurement* section for more details.

Project Options



Project Sample Rate/Volume Damping

Sample Rate

Sets the sample rate of the Virtual Project. If you want or need to use a sample rate which isn't supported by your sound card, use the varipitch feature, which can be set in the playback parameters dialogue. When loading Wave Projects with differing sample rates than the VIP, the resulting Objects are resampled

automatically to match the VIP's sample rate. This is a non-destructive real-time effect.

Volume

When combining multiple audio tracks with peaks at or near 0dB, it is very easy for the combined master output level to exceed 0dB. The Volume parameter provides a quick way to scale back the master output level, so as to avoid exceeding 0dB.

You can select from four different attenuation values by clicking on (selecting) one of the following radio buttons:

0dB: This option provides no Input Attenuation, and can be considered the "Off" setting.

-6dB: This option provides 6dB of Input Attenuation.

-12dB: This option provides 12dB of Input Attenuation.

-18dB: This option provides 18dB of Input Attenuation.

Theoretically, each track of a 4-track stereo project must be reduced by 2 bits (12 dB). In the real world, tracks are often not recorded or played back at the maximum level, allowing reductions to be made accordingly. Raising the volume reduction setting may, however, result in clipping.

Please keep in mind that Samplitude reduces the volume of Wave Projects to the setting defined in the Virtual Project (that uses the Wave Project). This allows you to switch between the VIP and Wave Project without the usual issue of volume differences between the project types. If a Wave Project is opened in isolation (without using the Wave Project in a VIP), Samplitude automatically uses the maximum volume playback level.

Snap/Grid

Read details about the Snap to Grid function in the "View Menu" chapter.

Snap to: When this box is checked, the Snap feature is turned on.

Objects: This option activates the Object snap.

Range activates the Range snap and enables you to use the currently marked Range as a snap basis by selecting Get current Range for snap. This can be used as a simplified form of the Grid/Bar snap. The principle is the same, the Range borders will become snap reference points and these points are continued in both directions in equal distances according to the Range length. Mark a Range with the smallest time fraction (e.g. 1/16 note) and you've got a 1/16 snap. Use this option when you want to snap to distinct time positions without changing the project tempo.

Grid/Bars: Activates a snap based on measures. You can enter the tempo in BPM and additional measure attributes in the *Bar/BPM area*. The tempo can also be detected from the current Range selection via "Get BPM from Current Range". You can define the number of beats that the current area selection corresponds to - e.g. 4, if a complete 4/4 measure has been marked. The program now automatically displays the tempo (BPM) for the marked area.

Partitions are used to define the number of snap positions per beat, i.e. a value of 4 = 1/16 notes in a 4/4 measure.

Grid/Frames: Activates the snap based on frame length. The exact frame length is dependent on the selected SMPTE format (Options > Synchronization Settings).

Bar/Beat: Here you set the meter (numerator/denominator), the tempo in beats per minute (BPM), and the timer resolution in peaks per quarter (clicks/pulses per quarter note – often shown as PPQN).

Snap Offset to project start: Lets you define a starting point for the snap and grid.

Use Snap Offset also for Grid: When this box is checked, the snap offset (if set) is applied to the grid.

Shortcut: *Ctrl+ Shift + R*

Recording

Lock Recorded Objects: This option locks the newly recorded Objects as soon as the recording is over. This is useful if you do not want to accidentally move the new Objects in a VIP.

Note: An Object can be unlocked by first selecting it (click on the Object in the lower half of the track), then selecting Object > Lock Objects > Unlock Objects.

Group Objects after Multirecord: The Objects, which are created while you record more than one channel at one time, will be grouped. You can also ungroup these Objects anytime you want.

Editing

Destructive Mode: Only available for Wave Projects (see "Wave Editing" chapter)

Auto Crossfade Mode: When performing a cut or moving an Object over another, a crossfade is automatically inserted.

CD Arrangement Mode: When you activate this menu item, Samplitude rearranges newly-inserted Objects, and inserts pauses between the Objects (conforming to the Red Book Standard).
Please refer to the *CD > CD Arrange Mode* section for more details.

AutoSave Mode:

This group of settings activates Samplitude's AutoSave mode for VIPs. This is a very useful tool which will automatically create backup copies of open VIP's, during a Samplitude session.

Samplitude will *not* interrupt playback or recording to instigate a backup! The AutoSave process is activated during brief periods of inactivity, and only backs up the actual VIP (usually a very small file).
The backup tool can be set to overwrite the open file, or to sequentially save copies of the open file. Samplitude will automatically name the backup VIPs, and place them in the project directory.
The save interval can also be set by the user. We cannot think of a reason why you would leave AutoSave Mode inactive, but you can do so, if you wish!

Bar/BPM
Here you set the meter (numerator/denominator), the tempo in beats per minute (BPM), and the timer resolution in peaks per quarter (clicks/pulses per quarter note – often shown as PPQN).

Track information

Please refer to chapter “Track Menu” -> Track information.

Synchronization active

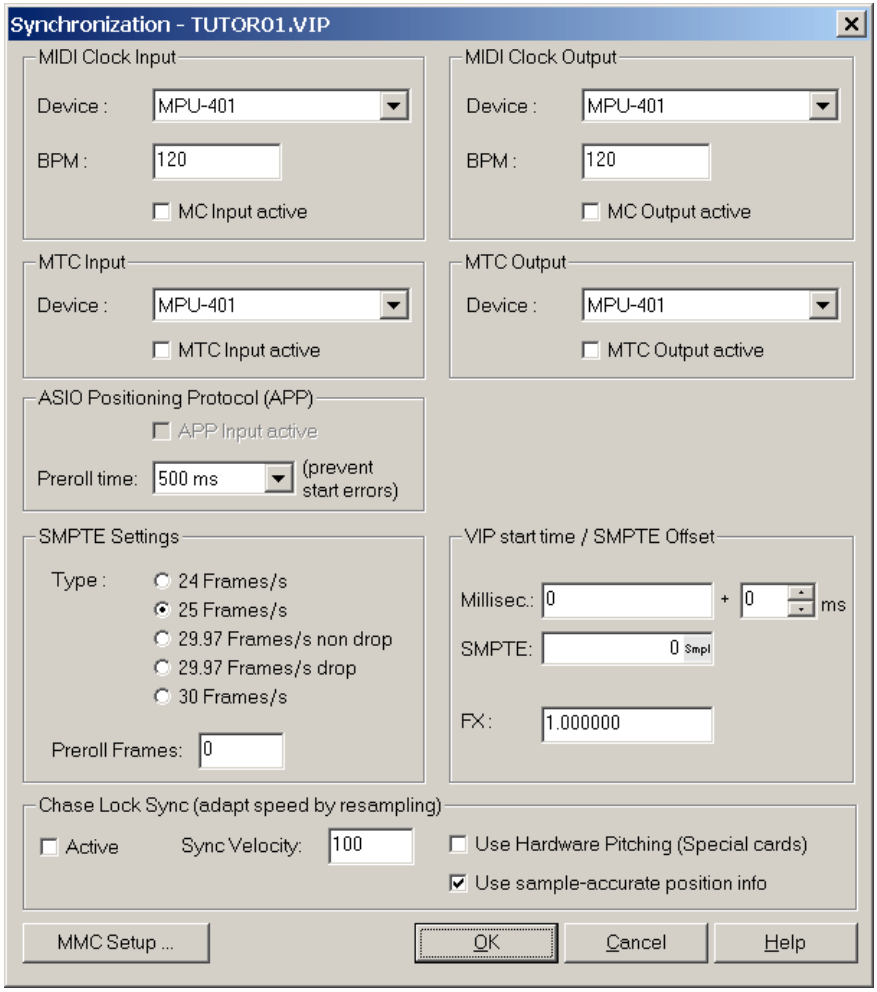
This menu item is mainly present to map a keyboard shortcut onto it. You can switch the Synchronization on/off via the SYNC Button on the transport control. Right-clicking this button opens the Synchronization Setup dialog

Shortcut: *“G”*

Synchronization Setup

Read the “Synchronization” chapter for an introduction on how to synchronize Samplitude with other programs or external MIDI devices.

Shortcut: *Shift + G*



MIDI Clock Input

If you want Samplitude to slave to MIDI Clock, use this section. This section tells Samplitude to look for MIDI Clock signals (start, stop, continue, song position pointer) on the specified MIDI input device. Set the section to "Active" if you wish to use this format.

Note: MIDI Clock does not carry any continuous timing information. It is a "Set and Forge" format, which does not allow Samplitude to monitor its lock with the external device. Use the SMPTE/MTC Input if you require extremely tight synchronization.

Device: Use the drop-down list to select the device that will receive MIDI Clock synchronization signals. This will be an active MIDI port (either virtual or hardware).

BPM: Enter the tempo, in beats per minute (BPM), of the audio which you want to synchronize with. This parameter doesn't affect the playback speed of the digital audio tracks, but is responsible for the grid display.

Active: Click on (i.e. enable) this option to activate synchronization to incoming MIDI Clock signals.

MIDI Clock Output
If you want Samplitude to generate MIDI Clock as the master, use this section. When active, Samplitude becomes the MIDI Clock Master, meaning that it will send out the following MIDI Clock signals: start, stop, continue, and song position pointer. No additional timing information is transmitted. This is often a quick solution for a synchronization setup which does not require tight sync between the components.

Device: Use the drop-down list to select the device that will send MIDI Clock synchronization signals. This will be an active MIDI port (either virtual or hardware).

BPM: Enter the tempo, in beats per minute (BPM), that Samplitude will send via MIDI Clock. This parameter will determine the tempo of the slave - i.e. If this parameter is set to 120, a sequencer slaved to Samplitude will play back at a tempo of 120 BPM.

Active: Click on (i.e. enable) this option to have Samplitude generate MIDI Clock Synchronization signals.

MTC Input
If you want Samplitude to slave to SMPTE/MTC, use this section. When active, Samplitude becomes a SMPTE/MTC slave device. It will listen for incoming time code signals, including timing information, on the specified MIDI Input device. If you require extremely tight sync, you should also activate the Chase Lock Sync option.

Note: Samplitude only understands MTC (the digital counterpart to SMPTE information). It is therefore necessary to utilize a MIDI interface that is capable of converting any possible SMPTE signals into MTC (MIDI Time Code).

Device: Use the drop-down list to select the device that will receive SMPTE/MTC synchronization signals. This will be an active MIDI port (either virtual or hardware).

Active: Click on (i.e. enable) this option to activate synchronization to incoming SMPTE/MTC signals.

MTC Output
If you want Samplitude to generate MTC as the master, use this section. Samplitude can also generate MTC signals, making it the master device in a Synchronization setup. MTC signals are sent to the specified output device whenever Samplitude starts/stops playback or recording.

Device: Use the drop-down list to select the device used for the transmission of MTC synchronization signals. This will be an active MIDI port (either virtual or hardware).

Active: Click on (i.e. enable) this option to have Samplitude generate MTC synchronization signals.

SMPTE Settings
These settings tell Samplitude what time format to use for synchronization. Different settings are available for various applications, such as the European and American Video- and Film formats.
Select the proper frame rate. Use 24 frames for cinematic synchronization, 25 frames for PAL video and audio synchronization, 30 frames for NTSC video. Note that the 29.97 Frames (drop and non drop) settings are also available!
Use the Preroll Frames option to allow for time code lock-up delays of external equipment.

Type: This parameter allows you to specify the SMPTE/MTC type used for synchronization.

Preroll Frames: This parameter allows you to specify the number of frames that Samplitude should ignore before synchronization occurs. Many devices need time to "lock up" to the synchronization signal, and this parameter allows you to compensate for this delay.

SMPTE Offset
This parameter allows you to specify a SMPTE offset time, in either milliseconds or SMPTE format. As an example, if you have a multi-track tape machine (as the master), striped with SMPTE, and you want your Samplitude project to start slaving to the tape machine at exactly 30 minutes, simply enter 30:00:00 in the SMPTE box.
The offset is removed from the incoming SMPTE time code signal to align the differences between the tape material and samples recorded in Samplitude. With an offset of '01:00:00:00' (1 hour), a tape that was previously striped can be synchronized if the recording/playback start point is set to 1 hour.

Millisec.: This numeric display shows the timecode offset in milliseconds.

SMPTE: This numeric display shows the SMPTE offset in SMPTE format (hours:minutes:seconds:frames).

FX: This option, similar to the Pitch (multiplication factor) parameter (in the Varipitch section of the Playback Parameter window), allows adjustments for small sync problems that can occur when dealing with long audio material. This feature alters the playback speed of the audio by the entered multiplication factor. Values above 1 cause an increase in playback speed. Values below 1 cause a decrease in playback speed.

Chase Lock Sync
Samplitude can perform Chase Lock Sync to incoming MTC signals. This is accomplished by manipulating the sample playback rate "on the fly" to adjust to fluctuations in the incoming MTC synchronization signals.

Note: This feature is extremely important when synchronizing to an analog tape machine, because its transport will introduce wow and flutter. If the timing changes are too big, Samplitude produces heavy pitch changes, which may be reduced by entering smaller Sync Velocity values in the Sync dialog (see below). Try values of 200 or more when you need fast pitch changes in Sync mode. The lower right status bar allows you to read the actual/maximum pitch change in cents (1 cent = 1/100 halftone).
Chase Lock Sync performs real-time resampling (if not using ARC44), which requires a certain amount of processing power. So be careful on slow machines!

Sync Velocity: This parameter allows you to specify how closely Samplitude follows fluctuations (in speed) of incoming MTC signals. In other words, this parameter will determine how quickly Samplitude follows a pitch change (caused by speed fluctuations in the MTC Synchronization signals), generated by the master device. The default value is 100, which will generally perform well. If you require a quicker response to pitch changes, increase this value to 200 or 300. If you require a slower response to pitch changes, decrease this value to 50 or 30.

Active: Click on (i.e. enable) this option to have Samplitude Chase Lock to incoming MTC synchronization signals.

Use Hardware Pitching (Special cards): Click on (i.e. enable) this option to allow the card's hardware to vary the sample playback rate. (If supported by your card)

Cancel: Click on this button to close the Synchronization window, without accepting changes to the settings.

OK: Click on this button to accept the current settings, and close the Synchronization window.

MMC Setup

See the "Synchronization" chapter.

Program Preferences

These items allow you to select a variety of settings to customize Samplitude to meet your working preferences and/or system.

Program Preferences → VIP Mouse Mode

The following options allow you to select from a variety of mouse modes. These modes determine the functionality of the mouse in VIPs.

Universal Mouse Mode

This option enables the Universal Mouse Tool. When using the Universal Mouse Tool, the upper half of a track is used to mark Ranges, and position the cursor. The lower half of a track is used to select and move Objects. Object handles, and volume and panorama curve events take precedence, and can be modified at any position within the track. The Universal Mouse Tool (default mouse tool) is the one you'll use most often.

Tip: You can also select the Universal Mouse Mode by clicking on (i.e. enabling) the Universal Mode icon in the upper toolbar.
Universal Mode Mouse Functions:

Right Mouse Button: Displays a context-sensitive pop-up menu.

Left Mouse Button:
1. Upper Track Pane:
Range: Selecting and moving Ranges (with Shift key).

Play Cursor: Single click sets the real-time cursor anywhere in the track.

2. Lower Track Pane:
Selecting Objects: Single click selects Objects (or group of Objects.)
Single click + Shift selects multiple Objects, including all Objects that fall between the two clicked Objects.
Single click + Ctrl selects multiple single Objects.

Moving Objects: Dragging moves Objects or Object groups.
Dragging + Shift moves Objects, or Object groups, up or down the track list, without changing the time position.

Duplicating Objects: Dragging + Ctrl duplicates one or more Objects.
Dragging + Shift + Ctrl duplicates one or more Objects, while allowing the Objects to be moved up or down the Track list, without changing the time position.

Object Editor: Double-click opens the Object Editor

Lasso Function: Clicking next to an Object on the empty track, while right-dragging the marquee (lasso) across multiple Objects, selects all Objects within the marquee.
Clicking next to an Object, while left-dragging the marquee (lasso), selects multiple volume or panorama curve events.

3. Complete Track
Object Handles: The upper handle changes the Object volume. The changed level is shown in dB in the upper left-hand corner of the VIP.
The upper left and right handles change the fade in/out of the Object. Fade curve parameters can be set in the Object Editor. The lower handles resize the Object.

Volume and Panorama Automation Curves: Single-clicking on the automation curve creates a new curve event. Double-clicking on the same curve event deletes the event.
Single-clicking on an event selects the automation curve event.
Single-click + Ctrl allows selection of multiple events.
Dragging a curve event moves the selected event.
Dragging + Ctrl moves several events.
To delete events, select them, and use Edit- > Delete .... Handle.

Range Mode (Secure Mode)

Select this option to enable the Range Mouse Tool. When using the Range Mouse Tool, the left mouse button can *only* be used to mark a Range. A Range can be marked no matter where (in the track) the mouse pointer is located.
Additional functionality is available through use of the following keys:

Period Key (.): If the period key is pressed, Samplitude temporarily changes to Object mode. Once in Object mode, you can easily select and manipulate Objects.

Minus Key (-): If this key is pressed, Samplitude temporarily changes to Curve mode. In Curve mode, you can change the volume, panorama and VST automation curves.
These special functions provide a quick way of reaching all important functions, while protecting an Object from accidental movements. This is why this mode is also called "Secure Mode".

Tip: You can also select the Range mode by clicking on (i.e. enabling) the Range mode icon in the upper toolbar.

Range Mode (Secure Mode) Mouse Functions:
Right Mouse Button: Displays context-sensitive pop-up menu.

Left Mouse Button:
Range: Selecting and moving Ranges (with the "Shift" key). Objects or automation curves can not be changed by accident (therefore called "Secure Mode").

Play Cursor: Single-click sets the play cursor anywhere in the track.

Select Object Mode temporarily: Using the (.) "period" key temporarily switches to Object mode, which allows Objects to be moved.

Select Curve Mode temporarily: Using the (-) dash key temporarily switches to Curve mode, which allows movements and manipulation of automation curve events.

Curve Mode

Select this option to enable the Curve Move and Grab Mouse Tool. When using the Curve Move and Grab Mouse Tool, the left mouse button can ONLY be used to create, delete, and move handles on custom volume or pan curves.

Tip: You can also select the Curve Move and Grab Mouse Tool by clicking on (i.e. enabling) the Curve Mode and Grab Mode icon in the upper toolbar.

Curve Mode Mouse Functions:
Right Mouse Button: Displays context sensitive pop-up menu.

Left Mouse Button:
Lasso Function: Dragging the marquee or lasso selects multiple volume or panorama automation curve events.

Volume and Panorama Automation Curves: Single-clicking on the automation curve creates a new curve event.
Double-clicking on a curve event deletes the event.
Single-clicking on an event selects the automation curve event.
Single-click + Ctrl allows selection of multiple events.
Dragging a curve event moves the selected event.
Single-click + Shift on a curve event selects all events between it and an already selected event. (Used to select a range of events)
To delete events, select the events and use Edit- > Delete .... Handle.

Object Mode

Select this option to enable the Object Mouse Tool. When using the Object Mouse Tool, the left Mouse button can *only* be used to select and move Objects. An Object can be selected and moved no matter where (in the track) the mouse pointer is located.

Tip: You can also select the Object Mouse Tool by clicking on (i.e. enabling) the OBJECT MODE icon in the upper toolbar.

Object Mode Mouse Functions:
Right Mouse Button: Displays context sensitive pop-up menu.

Left Mouse Button:
Selecting Objects: Single-click selects Objects (or group of Objects.)
Single-click + Shift selects multiple Objects, including all Objects between the two clicks.
Single-click + Ctrl selects multiple single Objects.

Moving Objects: Dragging moves Objects or Object groups.
Dragging + Shift moves Objects or Object groups up or down the track list without changing the time position. Dragging + "K" key moves the Object under the mouse pointer, as well as all Objects behind on the same track.

Duplicating Objects: Dragging + Ctrl duplicates one or more Objects.
Dragging + Shift + Ctrl duplicates one or more Objects, while allowing the Objects to be moved up or down the track list without changing the time position.

Object Editor: Double-click opens the Object Editor

Lasso Function: Clicking next to an Object while right-dragging the marquee (lasso) across multiple Objects selects the Objects within the marquee.
Clicking next to an Object while left-dragging the marquee (lasso) selects multiple volume or panorama curve events.

Object Handles: The upper handle changes the Object volume. The changed level is shown in dB in the upper left-hand corner of the VIP.
The upper left and right handles change the fade in and fade out of the Object. Fade curve parameters can be set in the Object Editor.
The lower handles resize the Object.

Object and Curve Mode

Select this option to enable the Object and Curve Mouse Mode. When using the Object and Curve Mouse Tool, the left mouse button can *only* be used to select and move Objects and select and move handles on custom volume or pan curves. An Object can be selected and moved no matter where (in the track) the mouse pointer is located.

Tip: You can also select the Object and Curve Mouse Tool by clicking on (enabling) the Object and Curve Mode icon in the upper toolbar.
Object and Curve Mode Mouse Functions:

Right Mouse Button: Displays context sensitive pop-up menu.

Left Mouse Button:
Selecting Objects: Single-click selects Objects (or group of Objects.)
Single click + Shift selects multiple Objects, including all Objects between the two clicks.
Single click + Ctrl selects multiple single Objects.

Moving Objects: Dragging moves Objects or Object groups.
Dragging + Shift moves Objects or Object groups up or down the track list without changing the time position.

Duplicating Objects: Dragging + Ctrl duplicates one or more Objects.
Dragging + Shift + Ctrl duplicates one or more Objects, while allowing the Objects to be moved up or down the Track list without changing the time position.

Object Editor: Double-click opens the Object Editor

Lasso Function: Clicking next to an Object while right-dragging the marquee (lasso) across multiple Objects selects the Objects within the marquee.
Clicking next to an Object while left-dragging the marquee (lasso) selects multiple volume or panorama curve events.

Object Handles: The upper handle changes the Object volume. The changed level is shown in dB in the upper left-hand corner of the VIP.
The upper left and right change the fade in and fade out of the Object. Fade curve parameters can be set in the Object Editor.
The lower handles resize the Object.

Volume and Panorama Automation Curves: Double-clicking on the automation curve creates a new curve event. Another double-click on the same curve event deletes the event.
Single-click on an event selects the automation curve event.
Single-click + Ctrl allows selection of multiple events.
Dragging of a curve event moves the selected event.
Dragging + Ctrl moves several events.
To delete events, select the events and use Edit -> Delete .... Handle.

Mouse Mode Studio 4.0

Right Mouse Button:
When using the Mouse Mode Studio 4.0 option (under File Menu > Preferences) and also restricting the mouse's function (Range Mode, Object and Curve Mode, Draw Volume, Draw Panorama, Object Mode, or Curve Mode), you can select this option to return the mouse function to normal. The right mouse button will select and move Objects, the left mouse button will mark Ranges.

Studio 4.0 Mouse Functions (Right Mouse Active):
Right Mouse Button:
Selecting Objects: Single-click selects Objects (or group of Objects.)
Single-click + Shift selects multiple Objects, including all Objects between the two clicks.
Single-click + Ctrl selects multiple single Objects.

Moving Objects: Dragging moves Objects or Object groups.
Dragging + Shift moves Objects or Object groups up or down the track list without changing the time position.
Dragging + key 'k' move the Object under the mouse pointer, plus all Objects behind on the same track.

Duplicating Objects: Dragging + Ctrl duplicates one or more Objects.
Dragging + Shift + Ctrl duplicates one or more Objects, while allowing the Objects to be moved up or down the track list without changing the time position.

Object Editor: Double-click opens the Object Editor

Lasso Function: Clicking next to an Object while right-dragging the marquee (lasso) across multiple Objects selects the Objects within the marquee.
Clicking next to an Object while left-dragging the marquee (lasso) selects multiple volume or panorama curve events.

Object Handles: The upper handle changes the Object volume. The changed level is shown in dB in the upper left-hand corner of the VIP.
The upper left and right handles change the Fade In and Fade Out of the Object. Fade curve parameters can be set in the Object Editor.
The lower handles resize the Object.

Volume and Panorama Automation Curves: Double-clicking on the automation curve creates a new curve event. Another double-click on the same curve event deletes the event.
Single-clicking on an event selects the automation curve event.
Single-click + Ctrl allows selection of multiple events.
Dragging of a curve event moves the selected event.
Dragging + Ctrl moves several events.
To delete events, select the events and use Edit -> Delete .... Handle.

Left Mouse Button:
Range: Selecting Ranges and moving Ranges (with "Shift" key).

Play Cursor: Single-click sets the play cursor anywhere in the track.

Zoom: Double-clicking outside of a selected Range (or no Range) zooms out.
Double-clicking inside a selected Range zooms in.

Open Wave Project: Double-clicking on a VIP Object opens the corresponding Range in the associated Wave Project.

Cut Mode

Select this option to enable the Cut Mode Mouse Tool. When using this mode in a VIP, the mouse pointer appears as a pair of scissors. Each click on an Object will split (cut) the Object.

Note: You can also easily cut (split) Objects by selecting an Object and pressing the "T" key.

Pitchshift/Timestretch Mode

Select this option to enable the Pitchshift/Timestretch Mouse Tool. This Mouse mode allows quick access to the real-time Object timestretching and pitchshifting

The lower right handle allows time compression or stretching of an Object by lengthening or shortening the Object. Using the middle handle, the pitch can be altered in a range of +/- 6 semitones.

Tip: You can also select the Pitchshift/Timestretch Mouse Tool by clicking on (i.e. enabling) the Pitchshift/Timestretch icon in the upper toolbar.

Draw Volume

Select this option to enable the Draw Volume Mouse Tool. When using the Draw Volume Mouse Tool, the left mouse button can *only* be used to draw custom volume curves. Notice that the mouse pointer appears as a pencil when located in a track. When you click and drag the mouse to draw a custom volume curve, the slower you move the mouse, the more handles are created.

Tip: You can also select the Draw Volume Mouse Tool by clicking on (i.e. enabling) the Draw Volume Mode icon in the upper toolbar.

Draw Panorama

Select this option to enable the Draw Panorama Mouse Tool. When using the Draw Panorama Mouse Tool, the left mouse button can *only* be used to draw custom pan curves. Notice that the mouse pointer appears as a pencil when located in a track. When you click and drag the mouse to draw a custom pan curve, the slower you move the mouse, the more handles are created.

Tip: You can also select the Draw Panorama Mouse Tool by clicking on (enabling) the Draw Panorama icon in the upper toolbar.

Scrub Mode

Select this option to enable the Scrub Mouse Tool. A speaker icon appears next to the mouse pointer – the left mouse button can now be used to scrub audio in real time by clicking/dragging in the desired scrub direction.

This functions as a pre-listen facility with control over play tempo. The project will be played forward or backward, with tempo controlled by the distance of the mouse pointer from the actual play cursor position. The larger the distance the faster the play tempo.

Tip: You can also select the Scrub Mouse Tool by clicking on (enabling) the Scrub icon in the upper toolbar.

Zoom Mode

Select this option to enable the Zoom Mouse Tool. In this mode, the mouse pointer becomes a magnifying glass which enables rapid horizontal zooms within the VIP. Left-clicking zooms in, right-clicking zooms out.

Tip: You can also select the Zoom Mouse Tool by clicking on (i.e. enabling) the zoom icon in the upper toolbar.

Program Preferences → Wave (HDP/RAP) Mouse Mode

Range

When working directly in Wave Project windows, this mode (the default mode) allows the left mouse button to mark Ranges.

Draw Wave

When working directly in Wave Project windows, this mode allows you to draw over existing waveforms (using the left mouse button), thus creating new audio. For example, you can use this mode to remove clicks. Simply zoom in on a click (it will appear as a spike), and draw a line across the bottom that connects its beginning and end points. In other words, you will be removing the click by drawing the audio without the spike.

Note: To use this mode, you must be zoomed in to a resolution where 1 pixel = 1 sample. You can quickly zoom in to this resolution by going to the View menu and selecting Horizontally > 1 Pixel = 1 Sample.

Draw Volume

When working directly in Wave Project windows, this mode allows you to adjust the height (volume) of a very small portion of the waveform. For example, you can use this mode to decrease (destructively) the volume of a plosive. Simply zoom in on a plosive, and click and drag the mouse from left to right (or vice versa) beneath the plosive. The height of the waveform will decrease, reflecting the volume change.

Tip: The position of the mouse pointer (above or below the waveform, when you click and drag), will determine whether the height (volume) of the waveform is increased or decreased and by what amount. The further the mouse pointer is above or below the waveform, the more radical the height (volume) change will be. The middle of the display is volume 100%, the lower border is volume 0%, and the upper corner is volume 200%.

Note: To use this mode, you must be zoomed-in to a resolution where 1 pixel = 1 sample. You can quickly zoom in to this resolution by going to the View menu and selecting Horizontally > 1 Pixel = 1 Sample.

Scrub Mode / Zoom Mode

These two options are functionally identical to the equivalent VIP Mouse Modes. Please refer to the VIP Mouse Modes section for further details.

Program Preferences → Object Mode

This feature is similar to the Group option (in the Object menu), but provides a much quicker way of selecting *all* of the Objects (either on a single track or *all* tracks).

Lock all Audio in Time: All audio tracks are locked in time.

Normal: If you've previously selected (i.e. enabled) any of the linking options, you can select this option to unlink *all* Objects. Multiple selection of Objects is possible.

Link Curves to Objects: Select this option and all custom volume and pan curves will be linked to their respective Objects. In other words, when you move an Object, the custom volume and pan and automation curves (above the Object) will move with it. This is especially helpful, if the curves are tailored to specific Objects and need to be moved with the Objects. Also in the other Object Link modes (see below) the curves are linked to the Objects. Multiple selection of Objects is possible.

Link until Silence: All Objects of a track, starting with the mouse position up to the next silence, are selected and moved together. This way, parts of a project – if separated by pauses – will stay unaffected by move operations made before the gap. Multi- selection of Objects is possible on different tracks.

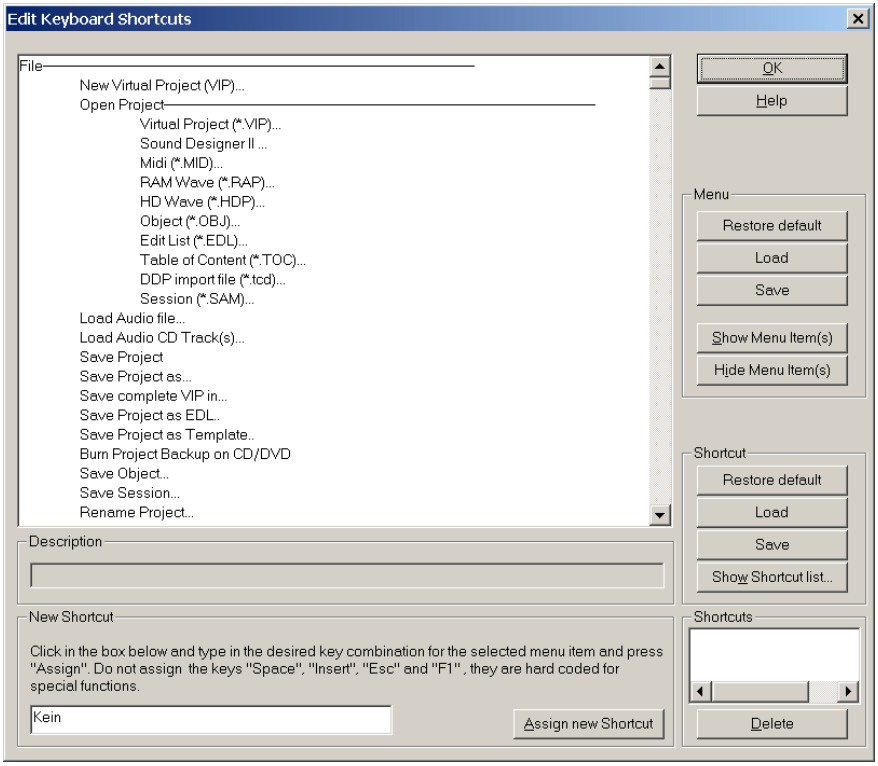
Link one Track: Select this option to quickly link *all* Objects in a track. For example, You can use this feature to quickly move *all* of the Objects (simultaneously), while preserving their relative time positions. Multi- selection of Objects is possible on different tracks.

Link all Tracks: Select this option to quickly link *all* Objects in *all* tracks. As an example, You can use this feature to quickly move *all* Objects in the VIP (simultaneously), while preserving their relative time positions. No multiple Object selection is possible.

Program Preferences → Edit Keyboard Shortcuts and Menu

Select this option to open the Edit Keyboard Shortcuts window. This window allows you to create your own shortcut keys for any of the menu options.

This makes it possible to assign your frequently used commands to specific keys and key combinations for quick recall.



The shortcut settings are saved to the file *Samplitude.ini* in the program directory when exiting the program. The next time the program is loaded, the customized shortcuts will be retained.

You can also customize the menus and hide menu items you infrequently or never use, e.g. import & export of unusual file formats or the surround functions. You can also save and load your keyboard shortcut and menus to/from a file. (*.syc; *.men)

The most important aspect of the dialog is the display of the Samplitude menu tree. You can select which menu option is to be reassigned a new key. To accomplish this, simply click on the desired menu option in the tree and then on the “New Shortcut” box.

Next, press the new key or key combination and press or click on the “Assign new Shortcut” button. Any possible combination of Shift, Alt, and Ctrl with other keys is possible.

Note: Please do not use the space bar, ESC, or the Insert key. These functions are hard-coded into Samplitude and cannot be changed!

In the upper left side of the window is a list of ALL Menu options. You can use the scroll bar to scroll through them. Any of the menu options can be selected (highlighted) by clicking on its name in the list.

Description: When you click on a menu option to select it, the function of the selected option will be shown here.

New Shortcut: After selecting a menu option, you can place the cursor in this box and type in the key (or combination of keys) that you want to use as a shortcut.

Assign Shortcut: After typing in the key (or combination of keys) that you want to use as a shortcut, click on the *Assign new Shortcut* button to assign the shortcut. You should now see the shortcut key (or keys) in the box above the "Delete" button.

Delete: After selecting a menu option, you can click on the "Delete" button to delete the shortcut key (or keys), thereby removing the shortcut.

Save Shortcuts: Click on the *Save* button to save your custom shortcuts. A standard Windows "Name File" dialog will appear. Name the file and click *OK*. Your shortcuts are now saved.

Load Shortcuts: Click on the *Load* button to load a previously saved "list" of custom shortcuts. A standard Windows "Open File" dialog will appear. Select (click on) the file containing the desired custom shortcuts, and click *OK*. The custom shortcuts will now be loaded.

Show Shortcut List: Click on the *Show Shortcut List* button and a list of assigned shortcuts window will open showing all assigned shortcuts.

Restore Default: Click on the *Restore Default* button to restore the "list" of shortcuts to their initial (default) settings.

Show Menu Items: The selected menu item in the menu item list will be shown in the Samplitude menus.

Hide Menu Items: The selected menu item in the menu item list will be hidden in the Samplitude menus.

Menu Items > Restore Default: This option restores the default menu view (all items will be shown)

Menu Items > Load/Save: These two buttons function identically to the "Load" and "Save" buttons for keyboard shortcuts (see above)

OK: Click on this button *(upper right corner of the window)* to accept the current settings and close the Edit Keyboard Shortcuts window.

Program Preferences → Change Toolbar Style

This menu switches between old (non-editable) and new (editable) toolbar style. Samplitude needs to be restarted for the change to take effect.

Program Preferences → Edit Toolbars

This dialog allows you to determine which toolbars will be visible in Samplitude. The dialog is simple to use – choose a toolbar from the available selection and either add or remove it from the current selection, then click close.
Individual toolbars and separators between toolbar sections can be moved using the Move Up/Down buttons. The Reset button resets all toolbars to the default view.
All toolbars can be floated and/or docked in accordance with standard Windows conventions.
The name/function of any toolbar can be easily discovered within the Samplitude workspace by hovering the mouse pointer above the toolbar – a tool tip will appear enclosing the toolbar name.

Shortcut: right-click directly on the toolbar

Program Preferences → Reset Toolbars

This menu item allows you to reset specific toolbar sections (Main, Positions, Punch, etc.) to the default view.

Program Preferences → Enable VIP Tool tips

This option enables Samplitude's informative tool tips. To disable the tool tips, uncheck this item.

Note: Although the tool tips are very informative and useful, they are drawn directly by the CPU (due to Windows design). There is therefore a very small overhead associated with using the tool tips. If your PC is underpowered, you may gain a small benefit by disabling tool tips.

Program Preferences → Grid Lines

This option allows you to select from 5 different grid display options when working with the grid. To switch off the grid display, select None.

Program Preferences → Video Height

If an AVI file is Linked to a Samplitude project, select this option to adjust the height of the AVI display. The different choices are: 25 pixels, 50 pixels, 100 pixels, and 200 pixels.

Program Preferences → Font Selection

Select this option to change the font that is used for general display purposes in Samplitude.

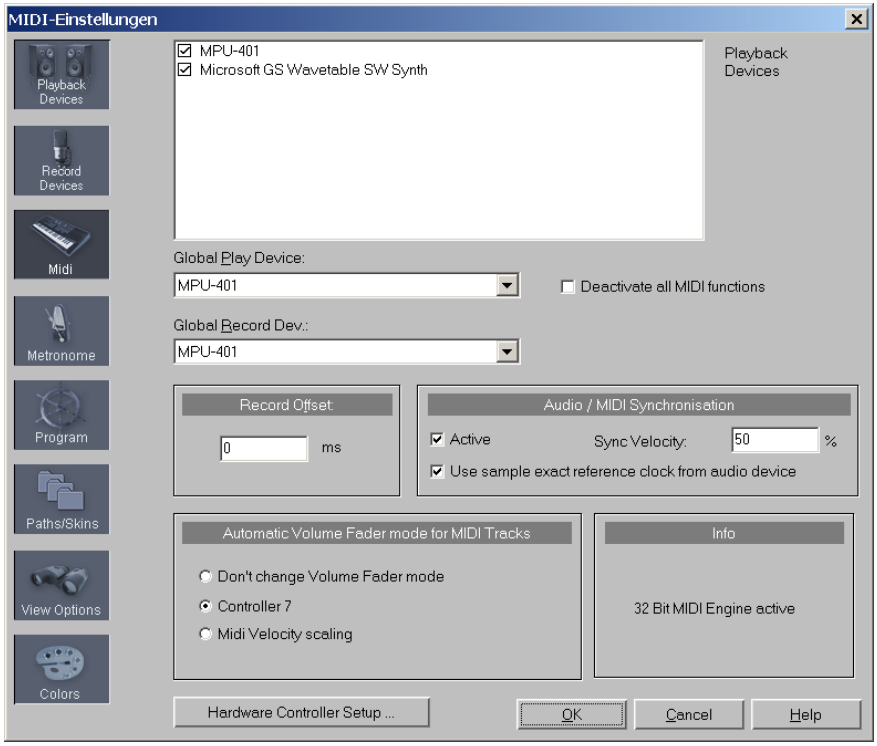
Note: If you notice words that are "cut off" when displayed, try using a different font or a smaller font size.

Program Preferences → Font for Time Display

Select this option to change the font that is used for the Time Display window in Samplitude. The Time Display window *must* be open or this option will have no affect. You can open the Time Display window by going to the Window Menu and selecting *Time Display*.

Note: The Time Display window can be resized, so the size of the font is not nearly as important as choosing a legible font.

Program Preferences → MIDI Options



The MIDI Options found in *File -> Preferences -> MIDI Options* and in the Track Info dialog allow you to set specific MIDI related parameters in Samplitude.

Playback Devices: This is the list of all installed MIDI devices in the system. Here you can hide unused MIDI devices from appearing in the input/output device list if you never intend to use them.

Global Play/Record Devices: These settings determine the global MIDI input and output devices - for any MIDI track in Samplitude that is not specifically assigned an input or output device. This can be considered the default setting for the MIDI playback and record device of your system.

Record Offset: Samplitude allows you to compensate for any delays that may be introduced by your MIDI setup. It may be necessary from time to time to adjust the Record Offset here if you notice slight offsets between your audio tracks and your MIDI tracks. If you do not notice any problems, leave the default value at 0 ms.

Audio/MIDI Synchronization: This setting determines how closely Samplitude matches the MIDI tracks with the audio tracks. On systems that have plenty of resources and faster CPUs, the values entered here can be adjusted close to 100% to force a perfect match. However, if your system seems to have problems coping with a higher setting, you may need to experiment with smaller values. The lower the percentage setting for the Sync Velocity, the more easily Samplitude matches the audio with the MIDI tracks.

Automatic volume fader mode for MIDI tracks

The track volume fader reaction can be changed manually at any time (right-click on the volume fader). When switching a track to MIDI or audio mode, however, the track-volume fader can be adjusted automatically. The automatic changeover to MIDI volume controller for MIDI tracks is activated by default. Select the required control mode you would like activated from a list when switching a track to MIDI mode (using the MIDI button). You can use controller 7 (MIDI volume) and MIDI velocity. Select "Don't Change Volume Fader mode" if you want to exclusively set the volume fader manually.

Note: When switching the audio mode back, the initial audio-volume assignment is restored.

Info: This section gives you information about what type of MIDI Engine is used to reproduce the MIDI tracks in Samplitude. Most systems will read 16bit. Emerging sequencing technology is just now introducing 32bit MIDI engines, which are a better match for pure 32bit environments such as Windows NT.
You can also disable all MIDI functions for troubleshooting purposes by checking the option "Deactivate all MIDI functions". Samplitude will then refrain from reproducing any MIDI data in any of its project windows.

Program Preferences → MIDI Options → Hardware Controller Setup

Samplitude makes it easy to tie in a hardware controller. We've prepared some presets for hardware controllers. You can change the presets or completely

reconfigure them. Take your time doing this, because it requires concentration. You will also require a MIDI interface (for controllers operating with MIDI) or a USB interface (for USB hardware controllers).

Is your hardware controller also listed in the Preset List?

To find out, you open the hardware controller Setup dialog. You can do this by pressing the "Y" key. Now click "MIDI Options". "MIDI controller Setup" opens the dialog.
Section "MIDI connection to control": In the highest flip menu you'll find a list with hardware controllers for which presets are available. Among them you will find the settings of the MIDI ports. USB hardware controllers usually offer their own MIDI ports. Controllers operated by a normal MIDI interface should have their MIDI port adjusted. It doesn't matter whether the equipment is specified in the Preset list.

Configuring Hardware Controllers/Creating your own sets

Should your equipment not be on the list or if you don't like some of the presets – don't worry. Samplitude is adaptable! In the "Controller" section (under equipment selection) you will find a flip menu with all remote control functions in Samplitude – each of which must assign the controller's messages. If you select an element (e.g. fader 1 for the first visible fader in the Samplitude mixer), activate the applicable element with your hardware controller and click the "Learn" button.

Note: Hardware controllers with touch-sensitive faders send messages to Samplitude when the fader is touched (or let go), meaning that the "Learn" button must be pressed after moving and before releasing a fader.

There's an explanation of the individual remote control parameters in Samplitude at the bottom of the page.
This is how to place each element of your hardware controller in Samplitude. If you've adjusted a finished set, save it under the name of the hardware controller. From now on it will always appear in the Presets list.
Note: You can use any other MIDI controller to control Samplitude (e.g. a MIDI fader box for individual track volume).

The following is a list of Samplitude parameters that can be remote controlled:

Key 1...8	Click: Turns mute on/off
Double click:	Switches solo on/off
in Shift Mode 1:	Automation for tracks
in Shift Mode 2:	Plug-ins in each track
in Shift Mode 3:	Link track pairs
in Shift Mode 4:	Automatic Record on/off
Link Master Volume	Turns the link button in the mixer master section on/off

Master Normalize	Normalize master
Shift Mode 1 Hold down:	Automation of tracks via keys 1...8 on/off
Shift Mode 2	Plug-ins on/off via keys 1...8 (calls up dialog when no plug-in is installed)
Shift Mode 3	Link tracks
Shift Mode 4 (Works as a Switch):	Automation record (Playback; Automation for the tracks must be activated) 1. Click: Displays fader movement; 2. Click: End of display Frequency regulation of the Master's 4 Equalizer bands using the fader for gain regulation 1. Click: Frequency regulator 2. Click: Back to gain regulator
Stop at position	Stop at current position
Play/Stop	Repeat from the current position/stop
Fader 1...8	Volume control of the first 8 visible channels in the mixer – or the Crossfade Editor Fader in Crossfade Editor Mode
Master Vol. left/right	Regulates the master volume left/right
Master EQ Lo/LM/HM/Hi Gain/Freq.	Controls gain of the master's 4 equalizer bands. In Shift Mode 4: Frequency regulation of the master's 4 equalizer bands
Master Stereo Enhancer	Regulates the multiband stereo enhancer in the master
Pitch Factor	Pitch Factor – used as a scrubber during playback. Changes are taken up even in Stop mode. The next playback shows the new pitch factor setting.
Mute 1...8	Switches the first 8 visible channels in the Samplitude mixer to mute
Solo 1...8	Switches the first 8 visible channels in the Samplitude mixer to solo
Record Ready 1...8	Channel path recording in the Samplitude mixer is set to "sharp".
Pot 1...8	Regulates the assigned value visible in the Samplitude mixer
Select 1...8	Selects VIP track.
Fadertouch 1...8 automation	Activates the recording of volume when automation of the individual tracks is activated. Only required for touch-sensitive faders.
Shift	Shift key for Hardware Controller key alternative.
Fader Master	Designates Master volume.

Scrub	Scrub mode on/off.
Scrubwheel FWD	Scrub forwards
Scrubwheel REW	Scrub backwards
Pot Touch 1...8	Press VPots on the Mackie Control
Play	Start Playback
Stop	Stop Playback
Fast FWD	Fast forward
Rewind	Rewind
Marker Mode:	Jumps to the next/previous Marker
Object Mode:	Selects the next/previous Object in the selected track.
Record	Starts recording immediately.
Mixertrack 1 UP	Scrolls 2 channels to the right.
Mixertrack 1 DOWN	Scrolls 2 channels to the left.
Mixertrack 8 UP	Scrolls 8 channel paths to the right.
Mixertrack 8 DOWN	Scrolls 2 channel paths to the left.

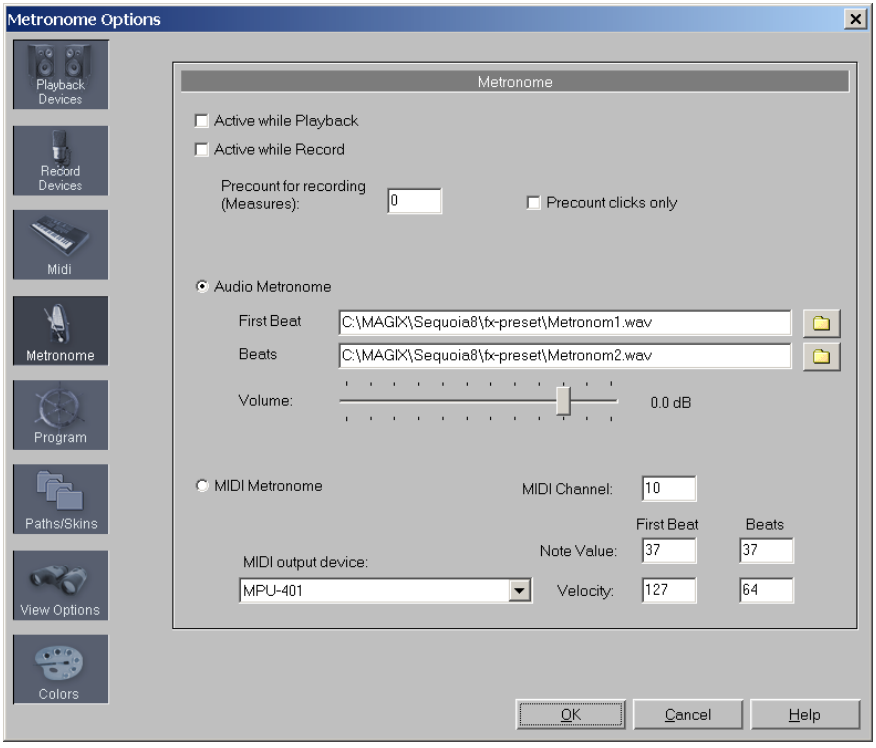
Channel path handling moves too. All channel paths in the mixer can be operated from the hardware controller.

Zoom	Cursor-controlled zoom (otherwise VIP is scrolled vertically and horizontally)
Zoom left	Vertical zoom out
Zoom right	Vertical zoom In
Zoom up	Horizontal zoom out
Zoom down	Horizontal zoom in (Only works when the Zoom key is activated)
Assig. Pan	The pots (VPots) 1...8 panorama is assigned.
Assig. Eq	The pots 1...8 Equalizer is assigned.
Assig. Aux	The pots 1...8 AUX-Sends level is assigned.
Marker 1...8	Sets markers 1-8. Play cursor jumps to the position of current marker.
Marker Mode	Switches to Marker Mode.
Loop On/Off	Loop on/off.
Enter	Confirms changes (OK/Enter)
Cancel	Cancels changes
Save	Saves open projects
Undo/Redo	Undo (Redo with "Shift" + "Undo/Redo")
Automation Read	REC/RDY-LED in MACKIE control – optical control of automation recording.
Automation	Signals volume automation curves in MACKIE control.
Automation Pan Touch	Records panorama automation when key is held down
Cursor left	Scrolls left in open project
Cursor right	Scrolls right in open project

Cursor up	Switches to the previous track in the current VIP
Cursor down	Switches to the next track in the current VIP
Mixer	Opens the mixer
CF Mode	Starts Samplitude Crossfade Editor
Cut	Cuts selected Object in two at the current cursor position
Option	Hold down keys F1-F4 on the Mackie control or one of the keys assigned by them (the F1-F4 corresponds) to switch between peak meter modes.

Program Settings → Metronome Settings

In this dialog, you can activate a metronome click which helps you play in time while recording. You can choose between a MIDI metronome (requires an external or internal MIDI sound generator) and an audio metronome, which then is played back via the mixer master channel.



Active while playback/record: You can activate the click for playback, recording or both.

Precount for recordings (measures): Enter the number of measures that the metronome counts in before the recording starts. If the option "Precount clicks only" is activated additionally, the click is muted after recording start.

Audio Metronome

First Beat/Beat: Choose samples for the first beat in the bar and the other beats in the bar. The sample for first beat should be a more "present" sample than the other one. Volume adjusts the overall volume of the metronome. With Device you can select the Output device.

Midi metronome

Here you can make settings for the MIDI metronome: The metronome then works by sending MIDI Notes, (usually on the drums channel), to an internal or external MIDI sound device, which must be present to hear any metronome click. You can assign the MIDI port for the output device, the channel and the note and velocity for the normal and the accented beat. Which beat is accented depends on the numerator setting in the Bars/Beats Dialog.

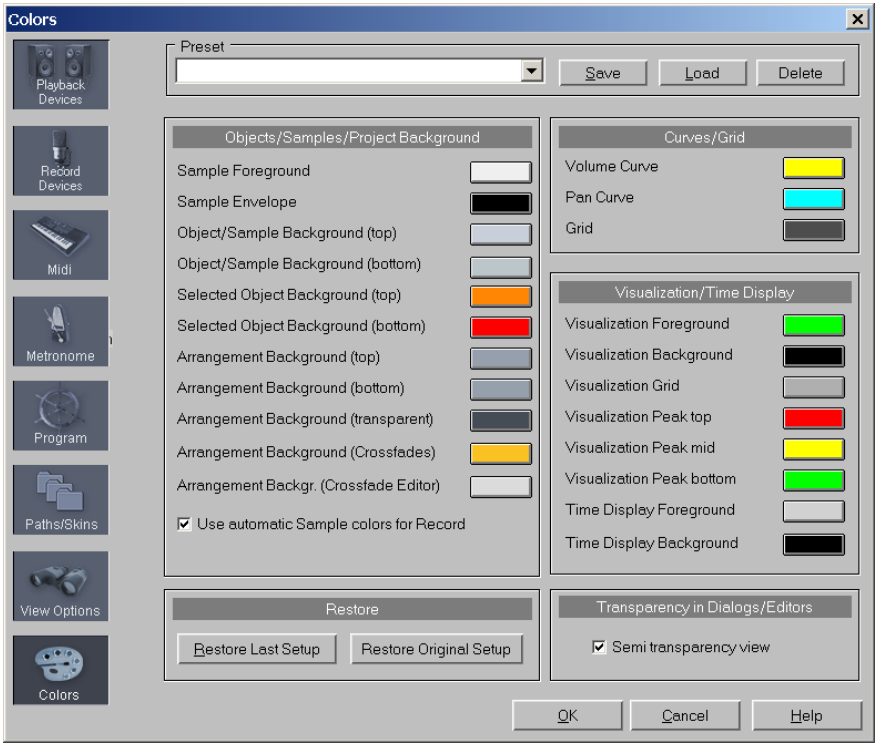
Program Preferences → Draw Setup

Please refer to "Menu View", "VIP Display Mode".

Shortcut: *Shift + Handle*

Program Preferences → Color Setup

Select this option to change the colors of various Samplitude displays.
You can select any color for options such as sample foreground and background, curves, VIP arrangement backgrounds, oscilloscope colors and so on.



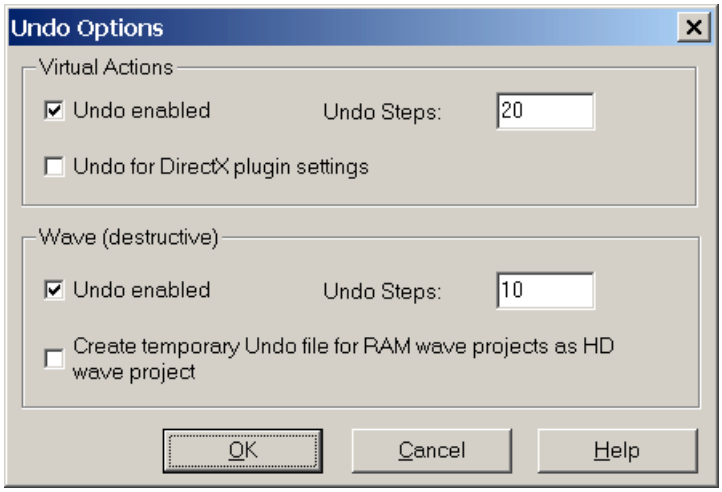
Samplitude's default colors have been carefully chosen to complement most working situations, but many variations are possible, to suit almost any taste. The best way to find out how changing the colors affects the working environment is to spend a little time experimenting.
Please remember that some variations are also available in the *View > VIP Display Mode > Definitions* menu item.

Note: Be careful when choosing your custom colors; it's possible to choose colors which will result in a screen that is unreadable. For example, if the text and background share the same color, you won't be able to see the information in the window.

The default color scheme can be restored at any time by choosing the Reset All Colors To Default option.

Program Preferences → Undo Definitions

Select this option and the Undo Options window will open. The Undo Options window allows you to specify the number of Undo Levels, and whether or not Undo is enabled.



The depth of undo can be specified when working with VIPs and Wave Projects. A value of "20" means that the last 20 changes can be undone.

Virtual Actions: Undo and the number of undo steps can be defined. It is also possible to disable undo step creation when changing DirectX plug-in settings. This will remarkably increase graphics performance when editing larger projects where many DirectX plug-ins are used.

Wave (Destructive): Undo and the number of undo steps can be defined. You can disable the creation of HDP-based undo files for RAM Projects for the sake of performance.

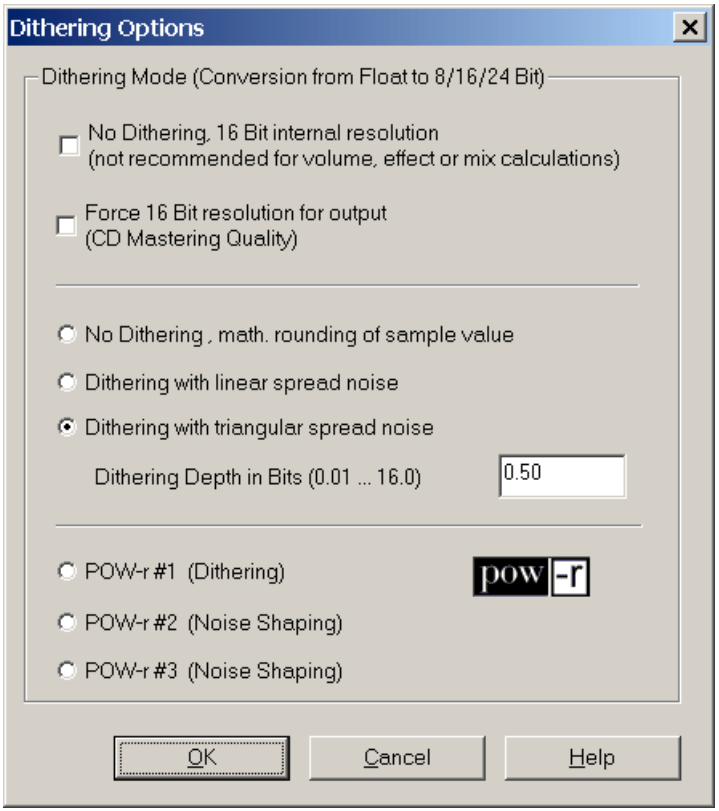
Cancel: Click on this button to close the Undo Options window without accepting changes to the settings.

OK: Click on this button to accept the current settings and close the Undo Options window.

Program Preferences → Dithering Options

Select this option to open the Dithering Options window. Dithering is used (in real time) when converting audio from higher resolutions to, e.g. 24bit to 16bit (often to burn a Red Book audio CD). The dithering process adds a very small amount of noise to the audio signal, which prevents low level audio from being distorted by quantization errors (truncation artifacts). The tiny bit

of noise introduced by this process is less distracting to the ear than quantization error, and is thus a good trade-off.



Dithering Mode (Real-time conversion from float to 16bit)

No dithering, 16bit internal resolution: All real-time calculations are carried out at an internal resolution of 16bit. Dithering is therefore not required. Compared to the usual internal resolution of 32bit, this achieves slight advantages of speed for slower CPUs. However, in this mode Samplitude is very susceptible to internal clipping. It should not be used for mixdown. One possible use is high-speed on-the-fly burning of MP3 files to a CD without effects processing. In this mode it is not possible to work in Surround formats.

No Dithering math rounding of sample value: This mode uses a mathematical rounding process to convert the audio; no dither is applied. Even without dither, this mode offers an audible advantage over simply truncating (lopping off) the extra bits.

Dithering with linear spread noise: This mode converts the audio using very low-level noise, which is spread over all sample values. The level of the noise can be adjusted with the Dithering Depth in Bits parameter (below).

Dithering with triangular spread noise: This mode converts the audio using very low-level noise, which is spread over all sample values in a triangular shape. This usually results in a more "decentralized" noise than linear dithering, leading to smoother results. The noise is not modulated by the signal, producing merging of a fading signal with the noise signal. The level of the noise can be adjusted with the Dithering Depth in Bits parameter (below).

Dithering depth in bits (0.01 - 16.0): This parameter allows you to adjust the level of noise (in bits) when applying dither. Generally speaking, values of 0.5 – 2.0 will yield good results. But, use your ears!

Tip: When applying dither, you want to use the least amount of noise necessary to prevent audible quantization artifacts.

POW-r Dithering

POW-r offers three distinct dithering modes.

POW-r #1: uses a special dithering curve to minimize quantization noise.

POW-r #2: (Noise Shaping): uses additional noise shaping over a wide frequency Range which can extend the dynamic Range by 5-10 dB.

POW-r #3: (Noise Shaping): uses additional, optimized noise shaping which can extend the dynamic Range by 20 dB within the 2-4kHz Range - the Range the human ear is most sensitive to.

Note: Noise shaping minimizes side effects caused by bit reduction, by spectrally displacing the quantization noise above 10 kHz - the Range the human ear is least sensitive to.

The mode which will sound best to you is primarily dependent on the program material and on your personal taste. The "right" setting is best determined by auditioning the audio material with each of the dithering modes.

How Does Dithering Work?

When an audio signal is quantized, the signal ends up with a "stair step" form based on the reduction of the possible amplitude values. This is the case for AD (Analog-to-Digital) conversion and when converting from a higher resolution to a lower resolution. An 8bit signal contains only 256 amplitude values. This type of stair step form leads to a deformity and therefore a distortion of the signal.
If a low level noise signal is added to the audio signal, the decay can be significantly reduced in respect of human ear perception.
Some excellent information regarding Dither can be found at *www.digido.com*
POW-R dithering stands for "Psychoacoustically Optimized Wordlength Reduction". For more information on POW-r, check out:
http://www.mil-media.com/docs/articles/powr.shtml

When is Dithering used in Samplitude?

Dithering is always used when converting from a 32bit float format to a 16bit format. This is the case for the following situations:

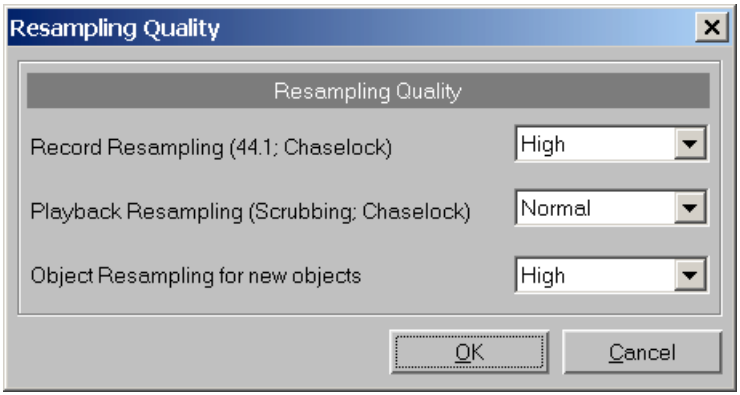
1. Playback of Virtual Projects on 16bit cards when using an internal precision setting of 32bit float.
2. Playback of high resolution Wave Projects on 16bit sound cards.
3. Creating audio CDs in real time using an internal precision setting of 32bit float.
4. Using track-bouncing of Virtual Projects that contain any resolution content when the internal precision is set to 32bit float and target format is less than that.
5. When converting any higher resolution Wave Projects to lower resolution Wave Projects.

Program Preferences → Advanced Buffer Settings

Please refer to "Menu Options", "System/Audio Settings", "Playback".

Resampling Options

You can set the resampling quality for several uses of resampling in the program.



Resampling can create aliasing. To avoid that in critical situations you can enable a high-quality resampling algorithm, but this also creates high CPU load, which is undesirable in many cases.
You can set the resampling quality for:
Recording: Resampling is used when recording from DAT (48 kHz sample rate) into a standard 44.1kHZ Wave Project. Default is high (should not be changed)

Playback: Used for Scrubbing and the varipitch function. Default is normal, because the resampled audio is only for listening purposes.

Object resampling for new Objects: Resampling is used when loading a Wave Project with a different sample rate in a VIP. Default is high. You can change the quality level in the Object editor -timestretching mode.

Program Preferences → Object Lock Definition

Use this dialog to set Object Lock Definitions. Please refer to the *Object > Lock Objects > Lock Definitions* menu item for more details.

Program Preferences → Set PreRoll Time

This option sets the pre-roll time for the cut simulation functions. (The cut simulation functions are located on the Punch-In toolbar.)
The pre-roll determines how far prior to the selected Range playback starts.

Program Preferences → Destructive effect calculation

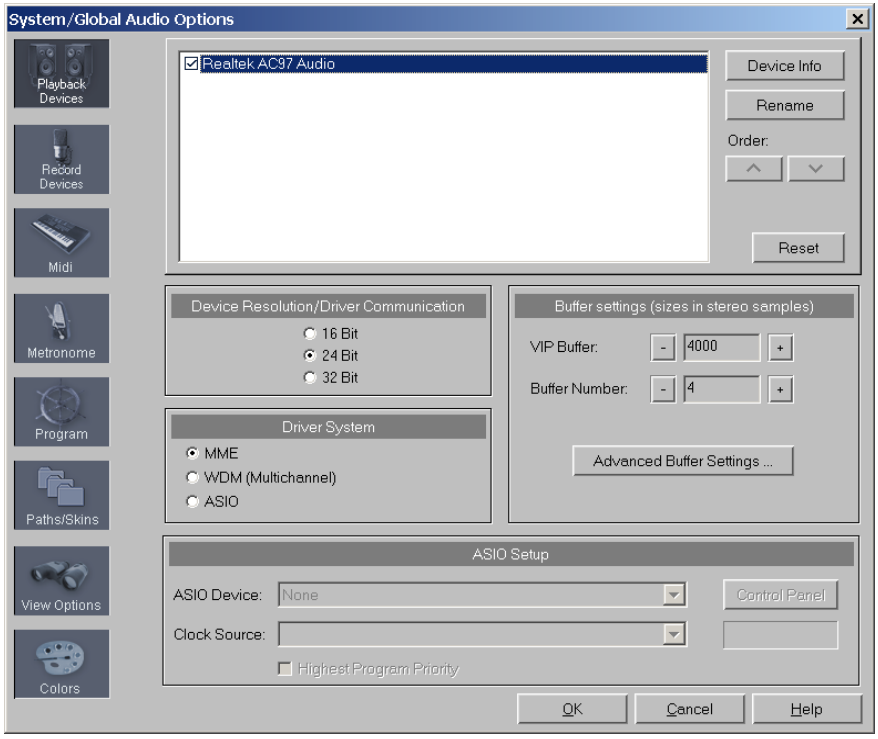
Please refer to the chapter "Offline effects" menu.

System/Audio

Use this option for direct access to several items of the "Options" menu.

Shortcut: *"Y"*

System/Audio → Playback Devices



The installed sound cards/sound card outputs are displayed in the large upper window. A small window is opened when pressing on the Device Window button. The new window displays information (supported sample rates, manufacturer, etc.) on the respective sound card in the adjacent window. Not required devices can be deactivated by unchecking the boxes. These are then no longer displayed in the device lists of the track properties and in the mixer. The order in which the outputs appear in the list can also be changed; the devices can be renamed.

Driver Resolution/Driver Communication
Select the bit resolution for addressing the audio driver during playback and recording. The selection of the resolution is usually performed automatically. If the output device cannot output a required resolution (e.g. 24bit), the corresponding lower resolution is then automatically created by Samplitude (dithering) and is passed on to the drivers. Under certain circumstances, e.g. if the driver generates error messages (because it reports certain resolutions, but cannot play/record these), you can force 16bit resolution.

Driver System
You can choose between MME, WDM Multichannel and ASIO. MME is Windows multimedia driver system. WDM (Multichannel): try to use this driver if your sound card supports 24/32bit playback and problems result when playing back high resolution audio files.

ASIO
ASIO drivers have some advantages over MME (provided that your sound card model is issued with ASIO drivers):
- Low latency of the driver system: This results in considerably lower response times for real-time processing. This enables software monitoring of the inputs (and VST instruments). The CPU load of the system drops, and more high-quality effects are possible without bouncing operations.
- Synchronization between recording and playback is carried out at the driver end. This ensures that the timing of overdub recordings is exa ct. This also enables mechanisms of compensating latency for effects that permit software effects monitoring.
- Furthermore, ASIO offers a general method for treating multi-channel audio. All bit resolutions and multi-channel problems that can otherwise occur with WDM drivers in Win2K/XP are thereby avoided fundamentally.
- Advanced hardware monitoring options are possible as a result of using ASIO Direct Monitoring (flexible routing inclunding pan and level).

ASIO Setup
ASIO Driver: The drivers of all ASIO Devices currently or previously available in the system are listed here. Control Panel opens the Setup dialog of the sound card driver. More information can be found in the sound card manual.

Buffer Settings

VIP Buffer: As error-free playback is usually more important than short response times, this value should be increased to about 8000 when playing back many tracks.

Buffer Number: You can state here how many of the above-described buffers are to be used. More buffers increase the safety, but also the memory requirement. The response times increase. The current buffer utilization during playback can be seen in the bottom right of the status bar.
These are the only relevant settings when playing Virtual Projects and when working with the mixer.

Advanced Buffer Settings

This window permits optimization of the buffer settings for VIP, hard disk caching and plug-in settings. No changes are usually required here. If you encounter problems just experiment with the provided presets.

Program Buffer (in Stereo Samples)
VIP Buffer: As error-free playback is usually more important than short response times, this value should be increased to about 8000 when playing back many tracks.

HD/Scrub Buffer: This buffer is used for playback directly in HD Wave Projects. You should also test lower values in terms of shorter response times!

Test Buffer: This buffer is only used for a real-time preview of effects.

Buffer Number: You can state here how many of the above-described buffers are to be used. More buffers increase the safety, but also the memory requirement. The response times increase. The current buffer utilization during playback can be seen in the bottom right of the status bar.

HD Performance (Track Speed)

Preview time for caching: You can state how many seconds of the VIP are to be preloaded into the cache.

Preload at Play Start: States the percentage of the cache that is loaded at play start. Higher values delay the start process, but also increase stability when playing back large track amounts.

Preload File Cache during Play Stop: When this option is activated and you position the play cursor, the file cache is always preloaded. In the status line the message "Cache Preloaded" appears. Playback is started really quickly since all the audio data required is already available in RAM.

Plug-in Buffer

The four numeric value entry fields define the buffer sizes of both plug-in types irrespective of each other as well as the resolution (in samples) for VST plug-in automation. All buffer sizes can have the same settings as the ASIO driver by entering "-1" as a value. Entering "0" states no limit for the buffer size.

DirectX preload buffer size: If no value is entered here, the entire buffer size is used for DirectX startup initialization.

System/Audio → Record Devices

The installed sound cards or available record devices are displayed in the large upper window. A small window is opened when pressing on the "Device Window" button. The new window displays information (supported sample rates, manufacturer, etc.) on the respective sound card in the adjacent window.

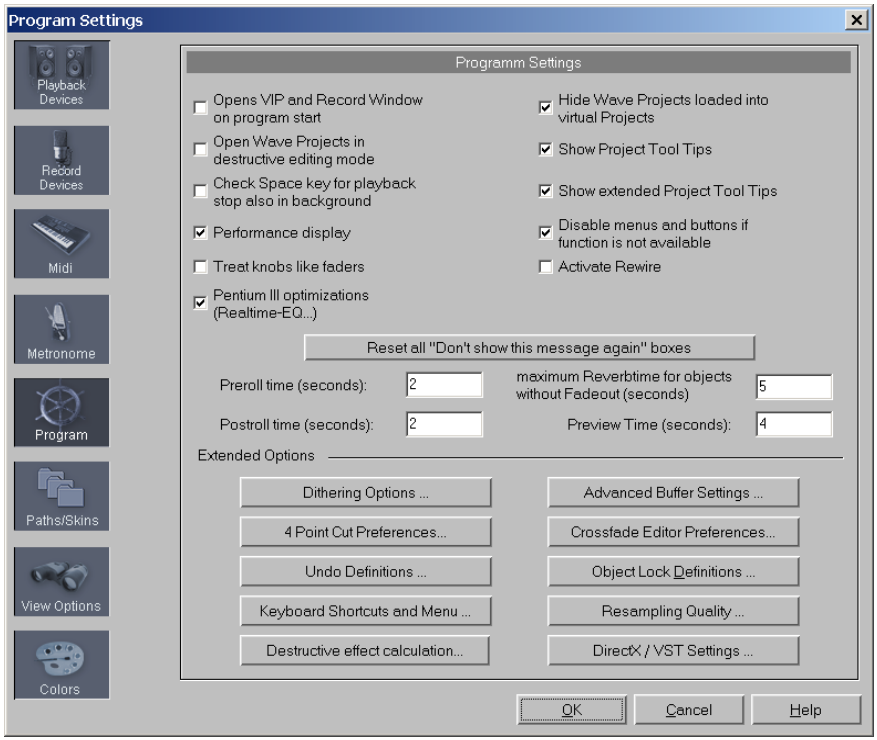
Monitoring Settings (Record Devices)

There are several monitoring possibilities. These settings are set for new projects. More info on monitoring can be found in the "Menu Play/Rec" chapter.

System/Audio → MIDI Options

Please refer to "Menu Options > Program Preferences > MIDI/Metronome Options".

System/Audio → Program Options



Opens VIP and Record Window on Program Start: Enable this option and Samplitude will automatically open a VIP and Record window when the program is started.

Open Wave Projects in destructive editing mode: Samplitude offers a non-destructive editing mode for Wave Projects which is pre-set for opening Wave Projects independently. Thus, all cut operations, effect settings, volume level changes, fades, and more are completely virtual in VIPs *and* in Wave Projects (WAV, RAP, HDP) - the audio material is not destroyed. This allows you to select the proper cut position, cut length, volume level, effect setting, etc. by experimenting without losing audio material or altering it. The "workspace" menu at the bottom left of the Wave Project screen allows you to select between "destructive editing" and "Wave editing" (non-destructive editing).
If you prefer the destructive editing, you can select this option for opening Wave Projects in destructive editing mode.
Note that opening Wave Projects directly from the VIP, so opening the Wave Project in the Object context, however results in working in Destructive Editing Mode.

Check Space Key for Playback Stop also in Background: Click on (i.e. enable) this option and Samplitude will continue checking the space bar (to stop playback) while multi-tasking in another program. If you wish to use the space bar in the other program (without stopping playback), do NOT enable this option.

Performance Display: This menu option allows you to enable the DSP power display in the lower left corner of the screen. Always be sure to use less than 100% to avoid the system hanging during playback.
If your project needs too much processing power, you can:
– Reduce the number of DSP effects in the Mixer or in the Object Editor.
– Reduce the number of tracks with audio Objects.
– Increase the VIP buffer size in Menu "System>Playback/Record Devices" to 16000 or 32000 samples.

The status bar will display suggestions to remedy problems if the processing capabilities of the CPU are exceeded (for example during use of the Noise Reduction or FFT Analyze Filter).

Show Project ToolTips: This switches Samplitude's ToolTips in VIP window on/off. You can choose between simple and extended ToolTips. Extended ToolTips show detailed information on the selected element and available keyboard shortcuts.

Disable Menus and Buttons if Function is not available: Click on (i.e. enable) this option and Samplitude will "gray out" functions that are not available. That is, some of the functions in the Effect menu are not available while working in the VIP window (they can only be applied destructively). In this case, the functions that are unavailable will appear "grayed out." If this option is not enabled, ALL functions will be listed "normally" whether they are available or not. It is recommended that you enable this option.

Pentium III Optimizations: Samplitude is designed to utilize the PIII algorithm extensions. This option is available on PIII-combatible CPUs istalled.

Preview Time (Seconds): This parameter allows you to specify the length of ALL calculated previews (in seconds). As an example, if you set this parameter to 10 seconds, when you click on the Preview button in the room simulator, Samplitude will calculate a 10 second preview of the results.

Treat knobs like faders: When active, you adjust the value of a knob by clicking on it and dragging the mouse up and down (as with faders). When inactive, you can "turn" a knob by dragging the mouse around it. For a finer resolution of the control, you can increase the radius of the circle.

Hide Wave Projects loaded into VIP: The windows of the Wave Projects loaded directly into the VIP are hidden by default (to maintain clarity on the

program desktop, because the program assumes that you want to work with them only inside the VIP). You can change this behavior here.
You can unhide all Wave Projects by "Window>Iconise all Wave Projects" and hide it again by "Window>Hide all Wave Projects".

Reset all "Don't show this message again" boxes: In a "fresh installed" state, Samplitude shows a lot of security questions at several places of the program. You can disable each of them by checking a little box named "Don't show this message again!" at the lower edge of the message box. If you want to see these messages again, press this button.

Preroll/Postroll time: When you perform cut simulations (play from/into cut start/end, play over cut in the "Playback" menu), then playback will start/end this time span before/after the chosen reference point.

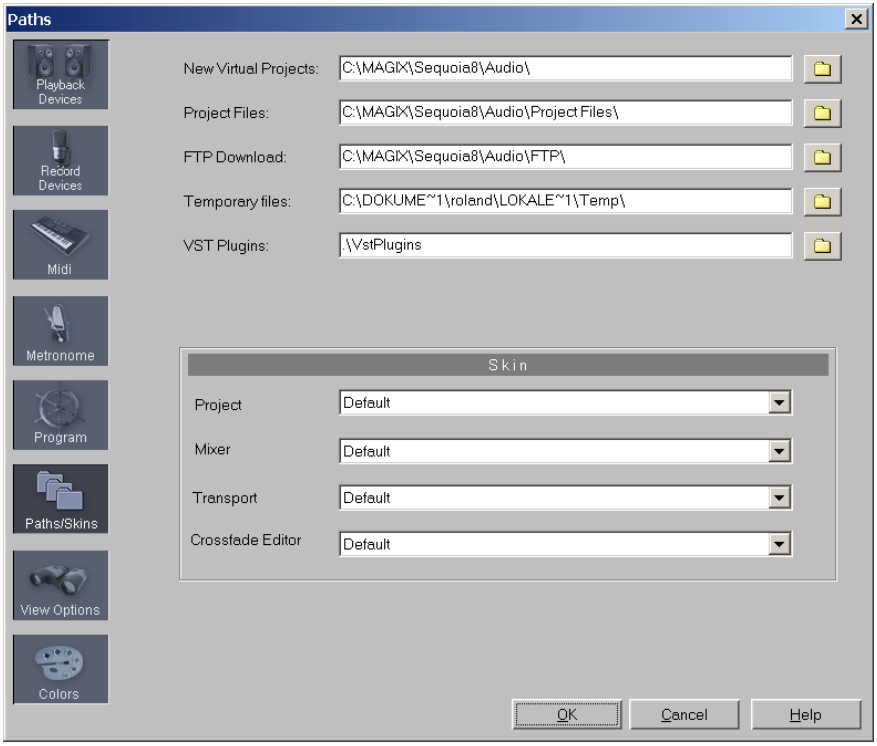
Maximum Reverb time for Objects without fadeouts: The time span the fx calculation will be continued after Object end to avoid cutting of reverb. Make sure that the Object volume is placed before the reverb in the effect chain or this will not work with Objects that have a fade out.

Preview time: The "Preview Time" setting determines the length of the preview function in various effect dialogs.

Dithering Options/Advanced Buffer Settings/Undo Definitions/Object Lock Definitions ...: You have direct access to this menu item. Please refer to the corresponding chapters of the menu "Options"!

OK: Click on this button to accept the current settings and close the System window.

System/Audio → Path/Skins



Paths

New Virtual Projects: Defines where all new VIPs and the newly recorded and imported wave files are saved.

Project Files: All other WAV files that are to be saved on the HD and which cannot be assigned to a concrete VIP are saved here.

FTP Download: All files downloaded via the integrated FTP client end up here.

Temporary files: Default folder for temporary files. This folder should be on a HD/partition with sufficient storage space.

VST-Plug-ins: defines the search path for VST Plug-in effects and VST instruments. At first start and if the VST path is valid, a VST scan is started when opening the track settings for the first time. If many VST plug-ins are installed, this can take several minutes. All plug-ins are then checked for usability in Samplitude. However, this scan is only required once. All used VST plug-ins are saved. If you have several VST plug-in folders, you can define a further folder that then only requires scanning once. Samplitude finds newly installed plug-ins automatically, but only at the stated location.

Skins

This window allows you to choose between different skins for the project, the mixer and the transport control. By using different skins you can change the colors and parts of Samplitude’s graphic display.
There are skins that are specially adapted to different work situations, such as the “Multi-Track Mixer”, which helps to maintain an overview in recording situations with many recording sources.
Further skins can be downloaded from http://www.samplitude.com
You can also choose between different skins at the left of the title bar in the System Menu of the corresponding window (Mixer, VIP).

System/Audio → View Options

Please refer to ”Menu View > VIP Display Mode”.

System/Audio → Colors

Please refer to ”Menu Options > Program Preferences > Color setup”.

”Window” Menu

Cascade

Choose this option to cascade all open windows.

Tile

Choose this option to tile all open windows.

Shortcut: Return

Untile

(shortcut key - Shift + Return) - If you previously chose the Tile option, you can choose this option to return all open windows to their original locations.

Arrange Icons

Choose this option to organize all icons (minimized windows) at the bottom of the screen.

Main Toolbar

Select (enable) this option to display the Main Toolbar. Unselect (disable) this option to hide the Main Toolbar.

Position bar

Select (enable) this option to display the Position bar. Unselect (disable) this option to hide the Position bar.

Punch/Play bar

Select (enable) this option to display the Punch/Play bar. Unselect (disable) this option to hide the Punch/Play bar.

Mouse Mode Toolbar

Select (enable) this option to display the Mouse Mode Toolbar. Unselect (disable) this option to hide the Mouse Mode Toolbar.

Range bar

Select (enable) this option to display the Range bar. Unselect (disable) this option to hide the Range bar.

Workspace bar

Workspaces should help you organize the huge amount of menu items and toolbar buttons to get a better overview over the functions of Samplitude. There are several items in the menus, toolbars and toolbar entries, that you never use, or only use at a certain point of your workflow.
The possibilities to hide menu items (menu ”Options>program preferences>Edit Keyboard shortcuts and menu....” and toolbars (menu ”Window”) and reorganize the contents of the toolbars (”Options>Program

preferences>Edit toolbars” or simply right-click a toolbar) are present in former versions of Samplitude. A workspace brings this all together and offers the ability to save such a setup under a certain name.
Several workspaces are predefined by the Samplitude team for different working tasks like recording, editing or CD mastering. The “Power User” workspace shows every toolbar and menu item. Use this if you are an experienced user or if you want to build your own custom workspace.

New workspace: Create a new workspace with ”New Workspace”. The currently chosen settings are copied to this workspace and all changes made from now on are stored in this workspace. You don’t have to save the workspace explicitly.

Edit workspace: The upper area allows you to select the toolbars to show, the middle area allows you to customize the toolbars, Edit menu calls the menu ”Options > program preferences > Edit Keyboard shortcuts and menu....” command.

Button bar

Shows or hides the Button bar on the lower part of the display. The buttons on the Button bar allow you to open/close the main windows of Samplitude.

Status bar

Select (i.e. enable) this option to display the Status bar. Unselect (i.e. disable) this option to hide the Status bar.

Mixer

Choose this option to open the Mixer window.

Shortcut: “M”

Time Display

Choose this option to open the Time Display window.

See chapter ”VIP window – a detailed look” for more details.

Visualization

Choose this option to open the Visualization window.

See chapter ”VIP window – a detailed look” for more details.

Transport Control

Choose this option to open the floating transport control.

See chapter ”VIP window – a detailed look” for more details.

Shortcut: *Shift + T*

Close all Windows

Choose this option to close all windows.

Shortcut: *"Ctrl + H"*

Iconise all Wave Projects

Choose this option to iconise (minimize) all HD Wave windows.

Hide all Wave Projects

Choose this option to hide all Wave Projects.

Half Height

Choose this option to reduce the program display to half height. This is useful when multitasking.

"Help" Menu

Help

Choose this option to open Samplitude's online help. By default, the online help will open with information about projects (HDP, RAP, and VIP) with an emphasis on the VIP window.

Help Index...

Choose this option to display a list of available help contents.

Context Help...

Choose this option to enable context based help. The mouse pointer will display a question mark, allowing you to click on any item or function to display relevant information.

Shortcut: *Shift + F1*

About Help...

Choose this option to display information on how to use online help.

Startup wizard/Tip of the day

Choose this option to display the "Start wizard" and "Tip of the Day" function.

About Samplitude...

Choose this option to display information about Samplitude. You can read your license serial number from here as well.

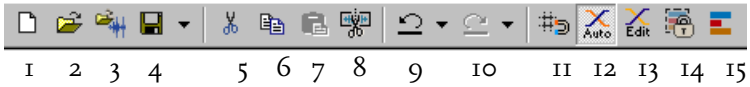
System Information...

A window is displayed which contains information about memory status and other parameters.
Particularly useful is the display of free storage on all connected disk drives, the system resources utilized by Samplitude and the memory usage. Make sure the parameter for system memory used by Samplitude never grows larger than the displayed overall system memory available (physical RAM). If this happens, the performance of Samplitude is reduced due to page-swapping (virtual memory) to compensate for the missing memory.

Button Overview

When the program is first started, only the tools and mouse mode bars are opened in the upper right corner of the monitor. All further ones may be added via the menu ”Windows”.
All toolbars may be placed on the screen at one’s discretion, and they are automatically arranged in the upper part of the screen by double-clicking on the header line.

Tools bar (left)



1 New multi-track project
2 Open VIP
3 Open WAV file
4 Export/Store project
5 Cut
6 Copy
7 Insert
8 Split Object
9 Undo
10 Redo
11 Snap on
12 Auto Crossfade Mode
13 Crossfade Editor
14 Group Objects
15 Ungroup Objects

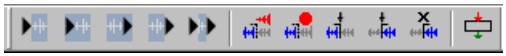
10 Redo
11 Snap on
12 Auto Crossfade Mode
13 Crossfade Editor
14 Group Objects
15 Ungroup Objects

Tools bar (right)

1 Set Marker
2 Set Track Index
3 Set Sub Index
4 Set Pause
5 Set CD End
6 Auto Track Markers
7 Make CD
8 Play Once
9 Play Loop

10 Play into Range
11 Stop
12 Record Options
13 Multi-I/O-Recording
14 Mixer

Punch/Playback bar



1 2 3 4 5 6 7 8 9 10 11

1 To Start Cut
2 From Start Cut
3 To Cut End
4 From Cut End
5 Beyond Cut
6 Punch-In Mode active
7 Punch-In Recording
8 Punch Start Marker
9 Punch End Marker
10 Delete Punch Marker
11 Live Input Mode active

Mouse Mode bar

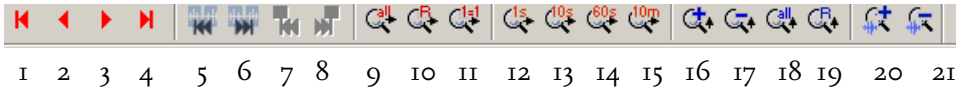
For mouse mode functionality see "Menu Options > Settings".

1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19

1 Universal Mode
2 Range Mode
3 Object Mode
4 Curve Edit Mode
5 Object and Curve Mode
6 Mouse Mode Samplitude 4.0
7 Cut Object Mode
8 Pitchshift/Timestretch Mouse Mode
9 Draw Volume Mode
10 Draw Panorama Mode
11 Draw Wave Freehand (only for Wave Projects)
12 Scrub Mouse mode
13 Zoom Mouse mode
14 Arrange all Objects
15 Normal Object mode
16 Link Curves to Objects
17 Link until silence
18 Link Objects on 1 track

19 Link Objects on all tracks

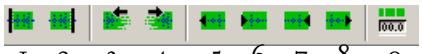
Position bar



1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21

1 – Cropping at start
2 – Half cropping to left
3 – Half cropping to right
4 – Cropping at end
5 – Object edge left
6 – Object edge right
7 – Range to last marker
8 – Range to next marker
9 – Zoom all
10 – Range as clipping
11 – 1:1 presentation
12-15 – Zooming Range 1 sec, 10 sec, 60 sec or 10 min
16 – Zoom in vertically
17 – Zoom out vertically
18 – Zoom all vertically
19 – Range as vertical clipping
20-21 – Waveform zoom in/out

Range bar



1 2 3 4 5 6 7 8 9

1 – Play cursor to start of Range
2 – Play cursor to end of Range
3 – Fold Range to the left
4 – Fold Range to the right
5 – Range start to previous zero-crossing
6 – Range start to next zero-crossing
7 – Range end to previous zero-crossing
8 – Range end to next zero-crossing
9 – Range editor

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