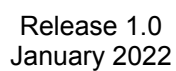


User's Manual



© Copyright 2022 by Stefano Daino

GRANT OF LICENSE. As registered user, the author grants to you the right to use one copy of the enclosed software program (the "SOFTWARE"). You may not network the SOFTWARE or otherwise use it or make it available for use on more than one computer at the same time.

DSP-AudioEditor COPYRIGHT. The SOFTWARE is owned by Stefano Daino, via Romeo Rodriguez Pereira 129/C - 00136 ROME - ITALY, and is protected by Italian copyright laws and international treaty provisions. Therefore, you must treat the SOFTWARE like any other copyrighted material (e.g. a book or musical recording) except that you may either (a) make one copy of the SOFTWARE solely for backup or archival purposes, or (b) transfer the SOFTWARE to a single hard disk provided you keep the original solely for backup or archival purposes. You may not copy the written materials accompanying the software.

OTHER RESTRICTIONS. You may not rent, lease, sell or transfer the SOFTWARE to other persons. You may not modify, adapt, translate, reverse engineer, decompile, or disassemble the SOFTWARE.

DISCLAIMER OF WARRANTY AND LIMITED WARRANTY. THE SOFTWARE AND ACCOMPANYING ELECTRONIC AND WRITTEN MATERIALS (INCLUDING INSTRUCTIONS FOR USE) ARE PROVIDED "AS IS" WITHOUT WARRANTY OF ANY KIND. FURTHER STEFANO DAINO DOES NOT WARRANT, OR MAKE ANY REPRESENTATIONS REGARDING THE USE, OR THE RESULTS OF THE USE, OF THE SOFTWARE OR WRITTEN MATERIALS IN TERMS OF CORRECTNESS, ACCURACY, RELIABILITY, CURRENTNESS, OR OTHERWISE. THE ABOVE ARE THE ONLY WARRANTIES OF ANY KIND, EITHER EXPRESSED OR IMPLIED, INCLUDING BUT NOT LIMITED TO THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE THAT ARE MADE BY STEFANO DAINO ON DSP-AudioEditor PRODUCT. NO ORAL OR WRITTEN INFORMATION OR ADVICE GIVEN BY THE AUTHOR, ITS DEALERS, DISTRIBUTORS, AGENTS, OR EMPLOYEES SHALL CREATE A WARRANTY OR IN ANY WAY INCREASE THE SCOPE OF THIS WARRANTY, AND YOU MAY NOT RELY ON SUCH INFORMATION OR ADVICE. THIS WARRANTY GIVES YOU SPECIFIC LEGAL RIGHTS. YOU MAY HAVE OTHER RIGHTS, WHICH VARY FROM STATE TO STATE. NEITHER THE AUTHOR NOR ANYONE ELSE WHO HAS BEEN INVOLVED WITH THE CREATION, PRODUCTION, OR DELIVERY OF THE SOFTWARE SHALL BE LIABLE FOR ANY DIRECT, INDIRECT, CONSEQUENTIAL, OR INCIDENTAL DAMAGES (INCLUDING DAMAGES FOR LOSS OF BUSINESS PROFITS, BUSINESS INTERRUPTION, LOSS OF BUSINESS INFORMATION, AND THE LIKE) ARISING OUT OF THE USE OR THE INABILITY TO USE THE SOFTWARE EVEN IF THE AUTHOR HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES. BECAUSE SOME STATES DO NOT ALLOW THE EXCLUSION OR LIMITATION OF LIABILITY FOR CONSEQUENTIAL OR INCIDENTAL DAMAGES OR THE LIMITATIONS OF DURATION OF IMPLIED WARRANTY, THE ABOVE LIMITATIONS MAY NOT APPLY TO YOU.

OTHER TRADEMARKS AND COPYRIGHTS:

- DSP-Quattro is Copyright © 2002-2022 by Stefano Daino. All rights reserved.
- D-SoundPRO is Copyright © 1998 by Stefano Daino. All rights reserved.
- Apple and all other copyrights and trademarks are held by their respective holders.

Index

INTRODUCTION	1
<i>What is DSP-AudioEditor?</i>	<i>1</i>
<i>About Third Parties AudioUnit Plug-Ins formats</i>	<i>2</i>
<i>What do I need to use DSP-AudioEditor?</i>	<i>2</i>
<i>What do I need to know to use DSP-AudioEditor?</i>	<i>2</i>
<i>Support:</i>	<i>2</i>
THE EDITOR WINDOW	3
<i>The worksheet</i>	<i>3</i>
<i>Each component has its own menu commands</i>	<i>4</i>
<i>The Editor Window is a container of documents, not a document by itself.</i>	<i>5</i>
THE DOCUMENT MANAGER.....	6
<i>Document Manager Commands.....</i>	<i>7</i>
<i>Using Drag&Drop to Import/Export audio files.....</i>	<i>9</i>
THE TRANSPORT CONTROLS	12
<i>The Transport Bar:.....</i>	<i>12</i>
<i>The Counter Display.....</i>	<i>12</i>
THE AUDIO EDITOR WAVEFORM VIEW	14
<i>The Channel Strip.....</i>	<i>16</i>
The Gain sliders, Peak and RMS VU-Meters and other controls	17
Using the Gain Sliders:	17
Using the numerical control above the Gain Slider:	17
Peak and RMS Meters:	17
The Panpot Rotary Control:	18
ON/OFF button:	18
Mute button:.....	18
Mono button:.....	18
L&R Phase Invert button:	18
<i>The Audio Overview view</i>	<i>19</i>
Using the Zoom Controller.....	19
<i>The Main Audio Editor Waveform view.....</i>	<i>20</i>
Making a waveform selection	22
Select All (or ⌘A):	23
Select Loop (or ⇧⌘A)	23
<i>The audio file Info side panel.....</i>	<i>23</i>
<i>Working with Markers</i>	<i>24</i>
Menu commands to move the playback cursor:	25
Go To Start:	26
Go To Next Marker (or ⌘K):	26
Go To Prev Marker (or ⇧⌘K):	26
Go To Selection Start:.....	26
Go To Selection End:.....	26
Go To Loop Start:	26
Go To Loop End:.....	26

The Generic markers table	26
The Generic markers commands	27
New At Playback (or ⌘M):	27
New At Selection Bounds (or ^⌘M):	28
New At Selection Start:	28
New At Selection End:	28
New At Loop Bounds (or ⌘M):	28
New At Loop Start:	28
New At Loop End:	28
Select All:	28
Unselect All:	28
Remove Selected (or ⌘M):	28
Remove All (or ^⌘M):	28
Lock Selected:	28
Unlock Selected:	28
Markers To Regions:	29
Selected Markers To Regions:	29
Markers List to New Project Audio Files:	29
Export Markers List As Audio Files...:	29
Copy Selected:	29
Copy All:	29
Paste:	29
<i>Working with Audio Regions</i>	30
The Audio Regions table	30
The Audio Regions commands	31
New From File (or ⌘U):	31
New From Selection (or ⌘U):	31
New From Loop:	31
Show All (or ^⌘U):	31
Hide All (or ^⌘U):	31
Remove Visible:	31
Remove All:	31
Region List to New Project Audio Files:	32
Export Region List to Audio Files...:	32
<i>Working with the Loop</i>	32
The Cycle Mode and the Loop	32
The Loop commands	33
Turn Loop ON/OFF (or ⌘L):	34
Lock Loop Length (or ⌘L):	34
Loop Selection (or ⌘L):	34
Show/Hide Loop Window (or ⌘L):	34
The Loop Window	34
EDITING AN AUDIO FILE	37
<i>Audio File Formats</i>	37
<i>Undo/Redo</i>	37
<i>Introducing the Real-Time Previewer Controller</i>	38
The Preset programmer:	38
The preview transport bar:	39
<i>The File Menu Commands</i>	40

New Audio File	40
New Empty Audio File (or ⌘N):	40
New Audio File from Selection:	40
New Audio File from Clipboard:	40
New Audio Files from Markers:	40
New Audio Files from Regions:	40
New Audio Files from L&R Channels:	41
Open Audio File	41
Import Audio File...	41
Import Audio File from Dual Mono L&R:	41
Import Audio File from Binary File:	41
Save	42
Save (or ⌘S):	42
Save As	42
Export Audio File As... (or ⌘E):	43
Export As iTunes Plus AAC:	45
Export Audio File As .aif:	45
Export Audio File As .wav:	45
Export Audio File As .mp3:	45
Export Audio File As FLAC:	45
Export Audio File As Dual Mono:	45
Export Selection As...:	45
Export Selection As iTunes Plus AAC:	45
Export Loop As...:	45
Export Markers List to Audio Files:	45
Export Regions List to Audio Files:	45
Save All	45
Save All:	45
Revert	46
Revert:	46
Duplicate Current Audio File	46
Duplicate Current Audio File... (or ⌘C):	46
Close	46
Close:	46
Close All	46
Close All:	46
<i>The Edit Menu Commands</i>	47
Cut Editor...	47
Cut	47
Cut (or ⌘X):	48
Cut Before:	48
Cut After:	48
Copy	49
Paste (Insert) and Replace	49
Delete	49
Insert Silence...	50
Append Clipboard...	50
Trim...	51
<i>The Commands Menu</i>	52
Rebuild Graphic Overview...	52
Zoom	52

Playback Cursor.....	53
Generic Markers.....	53
Regions.....	53
Turn Loop ON/OFF... (or ⌘L).....	54
Lock Loop Length... (or ⌘⇧⌘L).....	54
Show/Hide Loop Window (or ⌘⌘L):.....	54
Select All.....	54
Select Loop.....	54
The Processing Menu.....	55
AU Plug-in.....	55
Render Insert FX.....	56
Digital Effects.....	57
Digital Filters.....	57
Find Peak&RMS.....	58
Detect BPM.....	58
MonoToStereo.....	59
StereoToMono.....	60
Remove DC.....	60
Invert.....	60
Reverse.....	60
Swap L&R Channels.....	60
Normalize... (or ⌘G).....	60
Normalize To... (or ⌘⌘G).....	61
Apply Gain.....	61
Fade In... (or ⇧⌘F).....	62
Fade Out... (or ⇧⌘F).....	62
Fade In<->Out... (or ⌘⌘F).....	62
Fade Out<->In.....	62
X-Fade Looping... (or ^⌘L).....	63
Loop by BPM.....	63
Resample... (or ⌘⌘R).....	64
Speed&Pitch Change.....	65
Pitch Shifting.....	66
Time Stretching.....	66
THE CUT EDITOR TOOL	68
<i>Doing a graphical Cut using the Cut Editor Tool.....</i>	<i>68</i>
Cut... (or ⇧⌘X):	68
Setting fade-In/Out lengths and shapes of the curve.....	69
Trimming the the start and end of the region at cut point	70
<i>Inserting the Clipboard or Replacing with the Clipboard using the Cut Editor Tool</i>	<i>71</i>
Insert... (or ⇧⌘V):	71
Replace... (or ⇧⌘Y):	71
<i>Using the scissor mode to make additional cuts</i>	<i>71</i>
THE FADE EDITOR TOOL	73
THE X-FADE LOOPING EDITOR TOOL	74
THE BUILT-IN EFFECTS OF DSP-AUDIOEDITOR	76
Chorus	76

Stereoizer	77
Delay.....	77
Dual Delay	78
MultiTap Delay	79
Reverb 1 and Reverb 2.....	80
Parametric EQ	80
Graphic EQ	81
Resonant LowPass, BandPass, HighPass and BandReject Analog Filter emulations	82
The Gainer and Phase Inverter	83
External Audio Send&Return	83
THE INPUT/OUTPUT AUDIO RECORDER	85
<i>How to make a new recording.....</i>	<i>87</i>
<i>Rendering to file the audio processed by an external hardware unit.....</i>	<i>89</i>
PREFERENCES	91
<i>General Preferences</i>	<i>91</i>
<i>AudioEditor Preferences.....</i>	<i>93</i>
<i>Audio Settings</i>	<i>94</i>

Introduction

Welcome to the DSP-AudioEditor User's Guide. DSP-AudioEditor is the direct result of a very long experience developing Digital Audio applications for the MacOS, starting in '94 with D-SoundPRO, the powerful audio editor developed for Apple MacOS. DSP-AudioEditor introduces the new dimension of real-time into the world of audio editors, turning your computer into a real time workstation for music production, plug-in hosting and audio editing. All of the above through the smoothest interface you have ever seen.

DSP-AudioEditor is designed for the most professional users, however, thanks to the simple and intuitive interface, DSP-AudioEditor turns out to be the ideal product for whoever has no experience in the field and would like to enter in the world of digital audio.

This User's Guide will show you all there is to know about DSP-AudioEditor.

What is DSP-AudioEditor?

DSP-AudioEditor is an extremely powerful Audio Editor application with extended real-time and off-line support of Audio Unit (AU) Effect plug-ins for the Apple MacOS.

- DSP-AudioEditor is a real time audio editor.

DSP-AudioEditor is the right software for all people who want to record, create or edit their audio files using a Mac. With DSP-AudioEditor you can record audio sources, edit existing audio files, create new audio files as well. All these using various existing formats, such as WAVE, AIFF, CAF, mp3, m4a, iTunes Plus AAC, etc.

You will then be able to edit the new file starting with the basic Cut, Copy&Paste commands, to the most advanced algorithms, as the Time Stretching, Frequency and Pitch Shifting, Linear Phase Sample Rate Converter (SRC), and Digital Real Time Effects, like reverbs, stereo enhancers, graphic and parametric EQs, Analog Filter Simulators, and so on. And it is all built-in!

But not only that: DSP-AudioEditor is able to host standard third parties plug-ins as well. DSP-AudioEditor Editor allows you to use any plug-in exactly as any other internal edit function, offering also a real time preview of the result before to apply it.

And, if you will not like the result, you have not to worry: DSP-AudioEditor internal engine keeps track of all edits and you can Undo/Redo your edits without any limitation.

- DSP-AudioEditor is a tool for software and hardware samplers:

For all those among you that have already experienced music and both hardware and software samplers, DSP-AudioEditor is the perfect editing tool. In fact, DSP-AudioEditor offers not only a way of processing in floating point precision to satisfy even the more professional user, it offers also all the graphical tools for the sample-based editing of your audio files. You will be able to set loop points through a special loop editor window, and apply any internal or plug-in effect directly to your samples! DSP-AudioEditor guarantees the direct compatibility of audio files such as AIFF and WAVE, including the loop points information.

About Third Parties AudioUnit Plug-Ins formats

DSP-AudioEditor is capable to support Effects plug-ins. These plug-ins can be used as real time modifiers or as offline edit function with real time previewing. More about this will come later on this manual.

DSP-AudioEditor supports Effects Audio Unit (AU) 64 bits plug-ins.

What do I need to use DSP-AudioEditor?

DSP-AudioEditor works on any Apple Macintosh with an Intel or Apple Silicon (M1 or newer) microprocessors running MacOS 10.11 (El Capitan) or higher. It is a universal application, it runs natively also on Apple Silicon-based Macs.

The minimum requirements are: a CPU Intel (or M1 Apple Silicon), 4GB of RAM, and around 50 MBytes of free space on your HD.

The audio and graphic engines of DSP-AudioEditor run at 64 bits and are multithread. If present, DSP-AudioEditor will use the CPU multicores.

Please consider also that the performances of DSP-AudioEditor are heavily dependent by the number of plug-ins and effects that are used at the same time.

Mac standard audio interface can be used with DSP-AudioEditor by CoreAudio, which is built-in in the MacOS.

DSP-AudioEditor supports also all the external hardware audio interfaces if compatible with MacOS.

What do I need to know to use DSP-AudioEditor?

This User's Guide assumes that you know how to operate the Macintosh and are familiar with the basic Macintosh terminology such as point, click, drag & the MacOS file system and desktop. If you aren't familiar with these terms, you should refer to the Macintosh User's Guide to learn the basic operations of using the Macintosh. This User's Guide assumes also that you know what technologies like AudioUnit and, if you are interested to use external audio devices, what Core Audio are.

Support:

DSP-AudioEditor official home page is:

www.DSP-Quattro.com

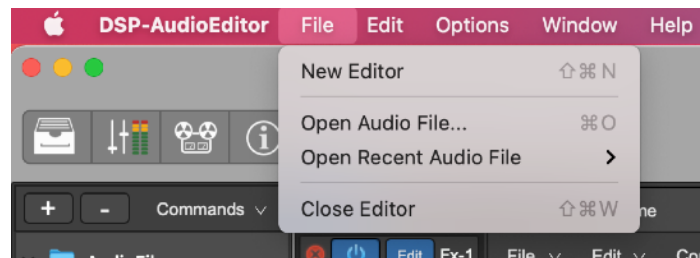
Here you can find additional information on our product.

At any time write us by e-mail at support@DSP-Quattro.com to have the answer to your technical questions or for more information regarding the use of DSP-AudioEditor.

The Editor window

The Editor window is the worksheet of DSP-AudioEditor. A worksheet is a container which includes all the objects you need to accomplish your work. At startup, DSP-AudioEditor automatically creates an empty Editor window. Then, you load your audio files on this Editor window to work on them. Closing an Editor window or quitting the application, DSP-AudioEditor asks if you wish to save or discard the edits which you have done on these audio files.

The command to create a new Editor window and to close it are on the top bar of the application, under the File menu:



where:

- **New Editor (or ⌘N):** to create a new empty Editor window and worksheet. DSP-AudioEditor will ask if you wish to save or discard your edits before to open a new empty worksheet, if any.
- **Close Editor (or ⌘W):** to close the current Editor window and worksheet. DSP-AudioEditor will first ask if you wish to save or discard your edits, if any.

On this menu there is also a command

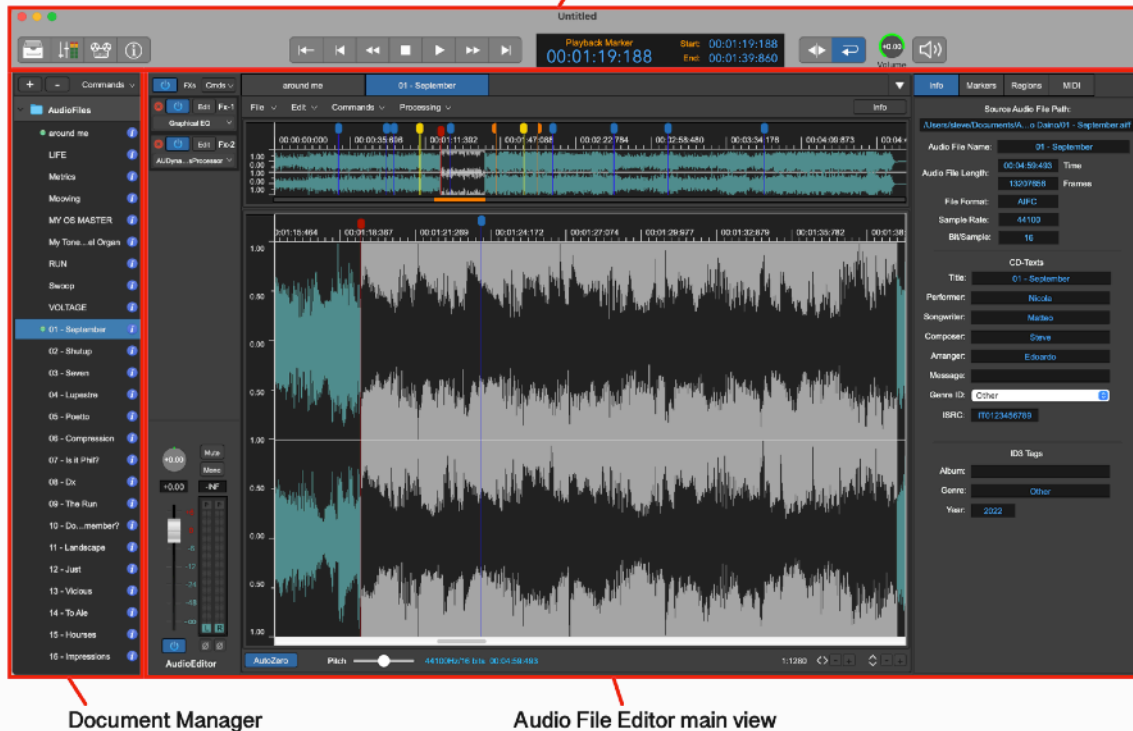
- **Open Audio File... (or ⌘O):** to open an audio file into a new Editor window if there is not yet an open Editor window - or on the current Editor window if there is already one open. DSP-AudioEditor will show a file browser to be able to choose one or more audio files to add on the current worksheet (more about this later in this manual).
- **Open Recent Audio File:** the application keeps track of the last used audio files. Using this menu, you can reload the most recently used audio files.

NOTE: it is not possible to have more than one Editor window open at the same time. For this, closing an Editor window and its worksheet, DSP-AudioEditor will first ask if you wish to save the edits made on current audio files, if any.

The worksheet

The Editor uses one main window, the worksheet. Even if DSP-AudioEditor also uses some additional accessory floating windows like the Loop tool or custom plug-in GUIs, most of the work is done on the main worksheet.

Transport Bar and Tool buttons



The worksheet consists of:

- **The Document Manager:** the Document Manager shows a list of the audio files which are open in DSP-AudioEditor.
- **The Audio File Editor main view:** it is the main view of the worksheet. It shows the waveform of the audio file under editing.
- **The Transport Bar and Tool buttons:** this view consists of the buttons to control the playback, the display to show information about current marker or playback cursor position and of the tool buttons to show/hide the views on the bottom part of the Editor window and additional tools and dialogs of the application.

More about these views in the following chapters of this manual.

NOTE: it is possible to show, hide, resize each one of the above components of the Project Document window. It is possible to resize their sub views as well. Just click and drag with the mouse the border of the view to resize it.

Each component has its own menu commands

The top application menu bar contains only the commands for the Editor window management and for the options which are related to the application, like the preferences and the audio settings.

Menu commands are at the top of the view where they apply. This means, as an example, that the commands to make an editing on an audio file are on top of the main Audio Editor view, the commands to manage the audio files loaded into the worksheet are on top of the Document Manager view, and so on.

Next chapters will deeply describe each view of the worksheet, as well as all the related menu commands.

The Editor Window is a container of documents, not a document by itself.

When you wish to edit your audio files, you open them into the Editor window. You must consider that it is not possible to save e reload an Editor as independent entity.

After editing an audio file, as soon as you will close it or when you will close the Editor window which contains it, you must choose to save it or discard the edits.

DSP-AudioEditor keeps track of the source audio file which are under editing. Saving the edits done on an audio files, the source audio file will be replaced by its edited version.

In other words, the audio files are the documents managed by DSP-AudioEditor, and not the Editor window. The last is not a document, but only a container of documents, the audio files.

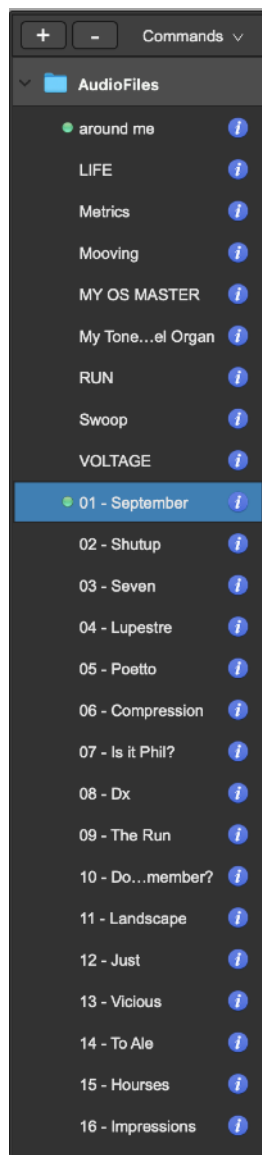
The Document Manager

On the left part of the worksheet, there is the Document Manager panel.

By default, when DSP-AudioEditor opens a new Editor window, the Document Manager panel is visible. Use the leftmost button on the top of the Editor window to show/hide the Document Manager panel:



To show/hide it, you can also drag the right border of the Document Manager panel view.



The Document Manager shows a list of the audio files which are used in DSP-AudioEditor at a certain moment.

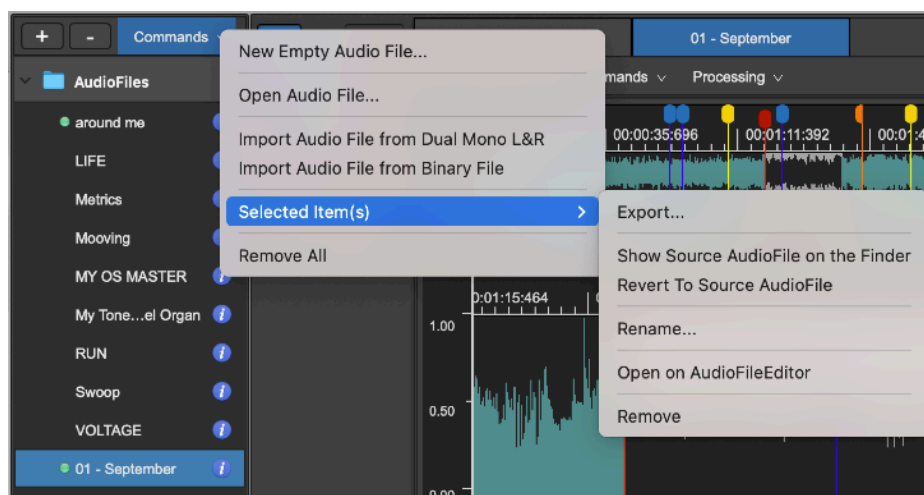
If the audio file has been already loaded on the audio file editor waveform view, a green dot on the left of the audio file name is visible.

For each audio file on the Document Manager table, there is a blue Info button on the right of the audio file name. Placing the mouse pointer on it, DSP-AudioEditor will show the file path of the source audio file. Clicking on it, DSP-AudioEditor will reveal the position on the MacOS finder of that audio file.

Document Manager Commands

In the upper part of the Document Manager panel there two buttons, one to Add a new element on the Document Manager table, one to Remove the current selected elements from the table.

Clicking on the **Commands** button on the top of the Document Manager table, it is possible to select one of the following commands:



- **New Audio File:** to create a new audio file. This audio file will be a temporary file until it is saved. After pasting some audio material on this, closing it, closing the Editor window or quitting the application, DSP-AudioEditor will ask to "save as" it, or to discard it.
- **Open Audio File...:** to load an audio file from file system. DSP-AudioEditor will show a file browser, allowing you to select an audio file to add to the Document Manager file table.
- **Import Audio File from Dual Mono L&R...:** to load a stereo audio file using two mono audio files, one for the left channel, one for the right channel (dual mono). DSP-AudioEditor will show a file browser asking to choose two audio files, then will add a reference to a temporary dual mono audio file to the file table.

- **Import Audio File from Binary File...:** to load a mono/stereo audio file from a raw binary audio file. DSP-AudioEditor will show an additional dialog to set the parameters which it needs to play correctly the file:

Bytes Order: Little Endian (PC)

Bit Depth: 16 Bits

Stereo/Mono: Stereo

Sample Rate: 44100 Hz

First Sample Offset: 0

Set

DDP Audio Binary File

where:

Bytes Order: can be LittleEndian (as used mostly on Windows and Intel and Silicon-based Macs) or BigEndian (as used mostly by old PowerPC-based Macs)

Bit Depth: 8, 16, 24, 32 or 32 floats - the bit depth of each sample on the binary file.

Stereo/Mono: set the number of interleaved audio channels.

Sample Rate: set the sample rate used to play the audio file

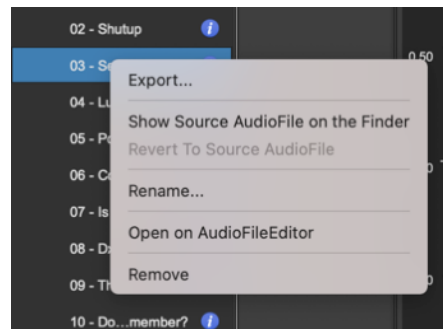
First Sample Offset: set an offset to start reading samples from the binary audio file. Set this value if the binary audio file has a header in front of the audio sample which DSP-AudioEditor must skip looking for samples.

On the right of this panel, there is a button to **Set** the current values to the values used by the DDP standard. Use these settings if you are importing the binary audio file generated exporting as DDP (Btw you should use the "import DDP" function of the AudioCD Layout to do it).

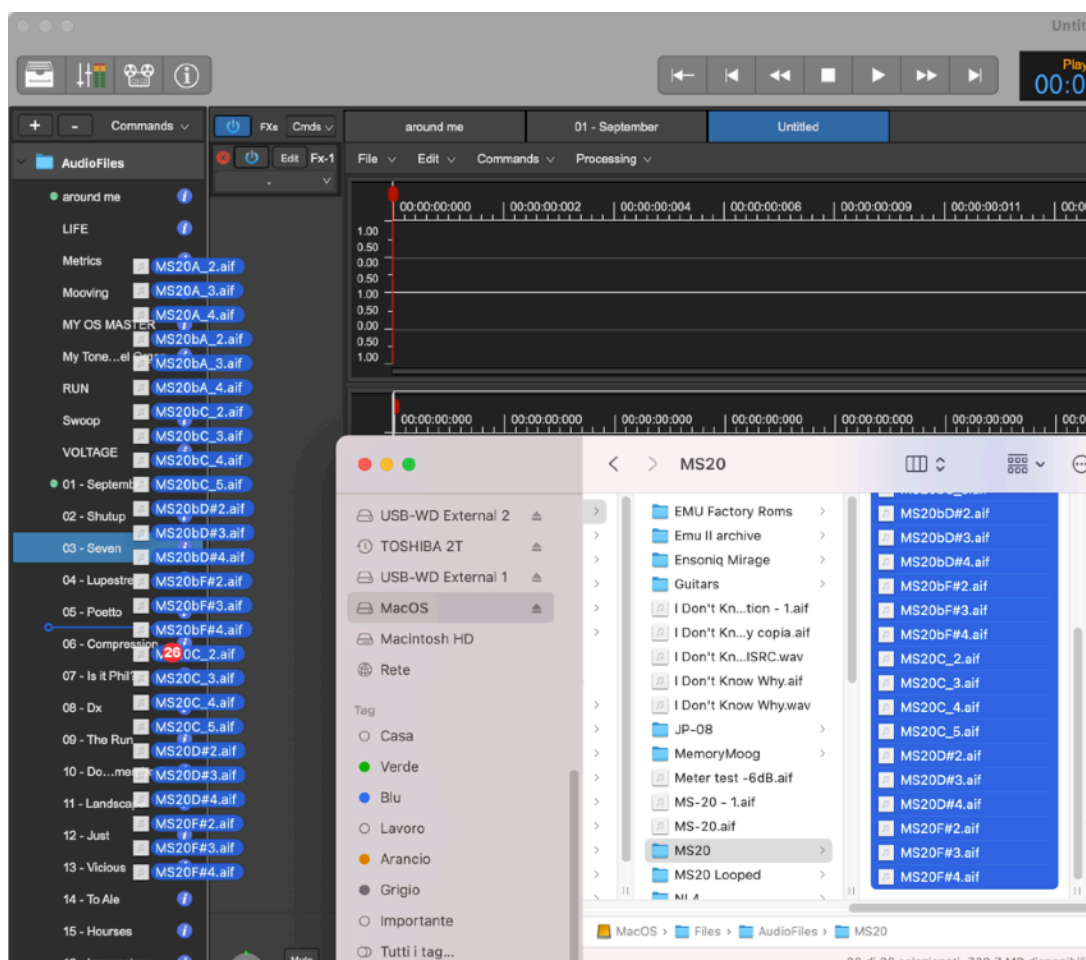
- **Select Items...:** the commands on the sub menu apply to the audio files which are currently selected on the table. It is possible to:
- **Export... :** it is equivalent to do a "Save As...". It is necessary to select an audio file on the table before to export it. Please refer to the chapter about File commands on the Audio Editor main view later in this manual for the details.
- **Show Source File on the Finder:** DSP-AudioEditor will reveal the position on the MacOS finder of that audio file.
- **Revert To the Source Audio File:** use this command to discard any changes made after the last save command.
- **Rename...:** to rename the current item selected item on the file table
- **Open on the Audio File Editor:** to load the selected item(s) on the audio file waveform editor and show it.
- **Remove:** this command will remove the selected item(s) on the file item. In case, before to remove the item(s), DSP-AudioEditor will ask to save or discard the changes, if any.

- **Remove All:** this command will remove all the items which are on the file table, empty it. In case, before to remove the item(s), DSP-AudioEditor will ask to save or discard the changes, if any.

It is also possible to click on a certain audio file on the file table while keeping the right button pressed to show a menu with a sub-set of the above commands:










Using Drag&Drop to Import/Export audio files



It is possible to drag&drop audio files between the Document Manager file table or the audio file waveform editor view and the MacOS file system, on both directions:

A - Dragging one or more audio files from the desktop or from a folder on the file table of the Document Manager will load the file audio in the current worksheet. The result is the same as using the command **Import Audio File**. If you wish then to open these audio files on the Audio Editor waveform view, double click on the new file names inserted on the Document Manager table.

B - Dragging one or more audio files from the Document Manager table to the desktop or to a folder, DSP-AudioEditor will export these audio files. If a file having the same name already exists (if, for example, you drag the file to the same folder of the source file), DSP-AudioEditor creates a new file having the same name of the source file with an increasing number added to

	To play forward at normal speed
	First click: to stop Second click: to move the playback cursor to the beginning of the file.
	To play forward and reverse at double speed.
	To move the playback cursor to the next or previous marker or - if any - to the beginning or end of the current selection or - if any - of the file.
	To toggle On/Off the Cycle Mode. If the Cycle Mode is On, the playback cursor, reaching the end of the portion to play, jumps at start continuing to play. If Off, it stops at end.
	To toggle On/Off the Audio Scrub Mode (slow motion mode). When slow motion is ON, the playback speed - on both directions - is controlled by the speed control, the slider which is positioned at the bottom of the waveform view.
	To toggle On/Off the preroll. If the preroll is ON, pressing the play button, DSP-AudioEditor starts to play 2 secs in advance respect the current playback position.

the end of the destination file name.

Once opened an audio file in an Audio File Editor view,

A - Dragging one or more audio files from the desktop or from a folder on the Audio Editor waveform view, DSP-AudioEditor imports the audio file into the current Project AND will open the audio file in a new Audio Editor waveform view tab.

B - On the Audio Editor waveform view, clicking on a waveform selection without moving the mouse, after around 2 seconds the mouse pointer will change to a hand. It is therefore possible to drag the selection from the Audio Editor to the desktop or to a folder. In this case, DSP-AudioEditor exports the selection in a new audio file with the same name of source file. If it already exists (if, for example, you drag the file to the source folder), DSP-AudioEditor creates a new file having the same name of the source file with an increasing number added to the end of the destination file name.

C - Moreover, dragging the selection over the Document Manager table instead of the MacOS file system, DSP-AudioEditor exports the selection into a temporary audio file which is internal to the current worksheet. It is possible to use this as shortcut to create new files trimming the portions of the audio to export.

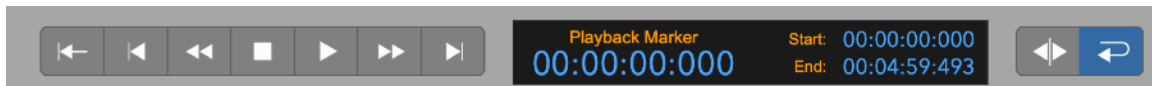
NOTE: it is even possible to drag the selection of only one channel of a stereo file to export only that channel (or a part of it).

D – The above points B) and C) work also if there is not a selected waveform on the Audio Editor waveform view. In this case, DSP-AudioEditor will export the audio file from the beginning up to the end. This feature is very useful as shortcut to the Audio Editor command 'Export Audio File As...', because it saves a copy of the current audio file as it is in that moment.

You are highly recommended to spend some moments making some practice with the extended Drag&Drop support in DSP-AudioEditor for exporting files. It is an irreplaceable tool to incredibly speed up your edits.

The Transport Controls

The transport controls are in the upper part of the Editor window. This section consists of the transport bar and the counter displaying the position of the current marker or playback cursor as well as loop and playback points.



The Transport Bar:

On the left part of the control section, there is the transport bar, which consists of the buttons to control the playback:

NOTE: Also the keyboard controls the playback:

- **Space Bar:** toggles Play/Stop. If it stops, the playback cursor returns to the position where the playback started.
- **Return Key:** to stop the playback.

At the bottom of the waveform view, there is the speed control, a slider which controls the playback speed:



When DSP-AudioEditor plays at normal speed, moving the speed control to the right causes the increase of the playback speed - as well as the pitch. Moving it to the left causes the decrease of the playback speed as well as of the pitch. Releasing the mouse, the speed control returns to center. Keep the CMD key pressed while dragging the speed control to lock it at the last position when releasing the mouse.

If the Audio Scrub button on the transport bar is ON, DSP-AudioEditor plays in slow motion. In this case the speed control works as jog-shuttle. If it is centered, the playback is on stop (playback speed = 0). Moving the speed control to the right, DSP-AudioEditor will play in slow motion, forward. Moving it to the left DSP-AudioEditor will play in slow motion, backward.

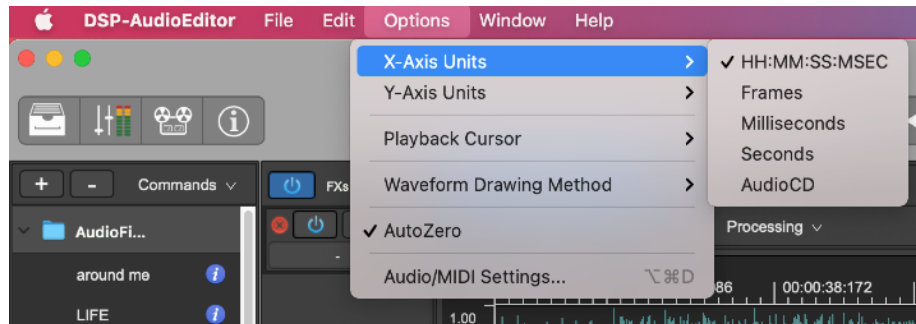
Using the ribbon controller, DSP-AudioEditor plays the audio file at a different pitch and speed. Raising the pitch implies a faster speed and shorter time duration, and vice versa. It is also possible to change the pitch without modifying also the speed - and time length - just using the Pitch Shifting processing function on the Processing menu. This processing function will be explained later in this manual.

The Counter Display

At the center there is the Counter Display, which shows the position of the playback cursor or of the current marker.



The Counter Display shows the position of the playback cursor in the main part. In the two smaller sections on the right, there are the positions where DSP-AudioEditor currently cycles playing the file. Clicking on any marker on the waveform editor view, the information on the Counter Display - name and position – will change accordingly. Clicking and dragging the mouse on the waveform view and making a waveform selection, the display shows the length of the new selection on the main part, while the counters on the right show the selection start and end.



The default time unit is hours:minutes:seconds:milliseconds (HH:MM:SS:MSEC). Possible choices are Frames, Seconds, Milliseconds or AudioCD (CDDA).

Use the menu command

Options->X-Axis Unit->...

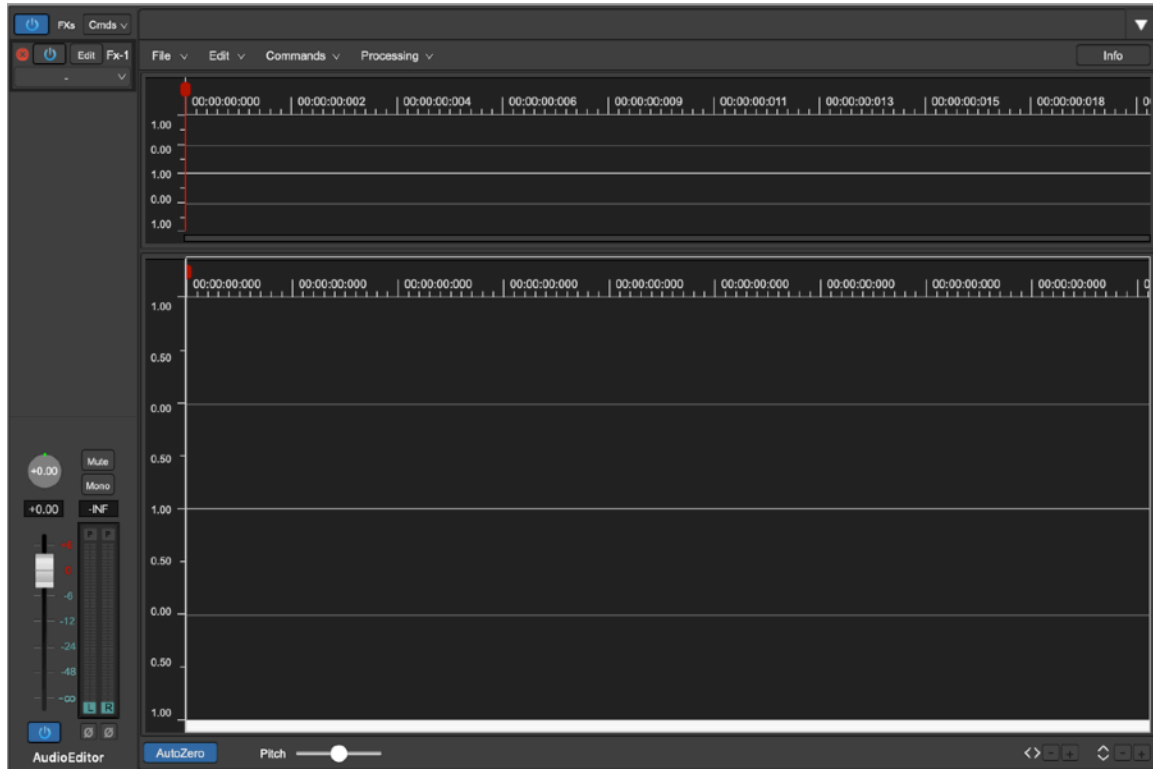
which is on the application menu bar

to make your selection.

On DSP-AudioEditor Preferences, it is possible to enter the default value for X-Axis units used by DSP-AudioEditor creating new Editor windows.

The Audio Editor waveform view

If you have not already loaded an audio file into the AudioEditor, DSP-AudioEditor will show an empty AudioEditor view:



To edit an audio file you need to load it first into the Project. This can be done by a two steps procedure, or by using a single command of DSP-AudioEditor, it will do both steps one after the other automatically.

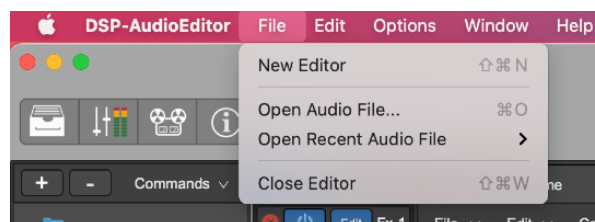
1 - Two Step procedure:

first load an audio file into the Document Manager, then open it into the AudioEditor doing one of the following actions:

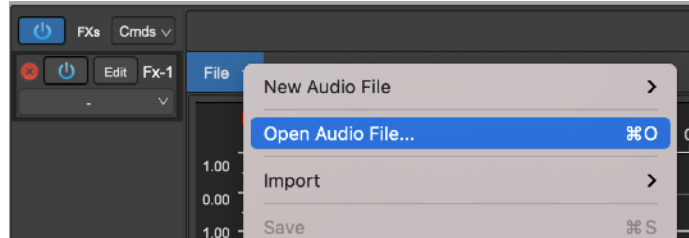
- double-click on the audio file name on the audio file table
- drag the file name from the audio file table to the Audio Editor waveform view.

2 - SingleStep procedure:

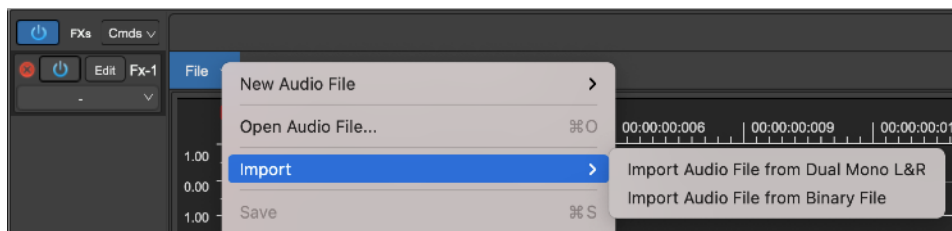
- use the menu command **File->Open Audio File...** on the top application menu bar:



- use the command on the popup menu control **File->Open Audio File...** on the top of the AudioEditor main view:

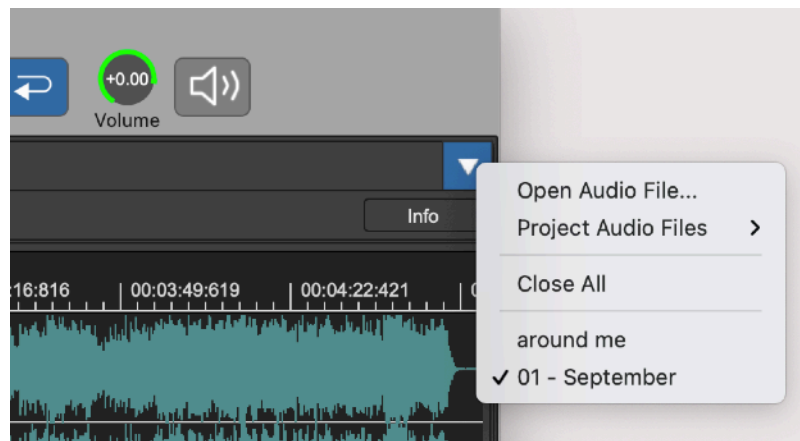


NOTE: it is also possible to use the commands on the **File->Import** sub menu to import a dual-mono and a Raw data audio file:



Please refer to the chapter about the Document Manager commands for a detailed explanation of these commands to Import.

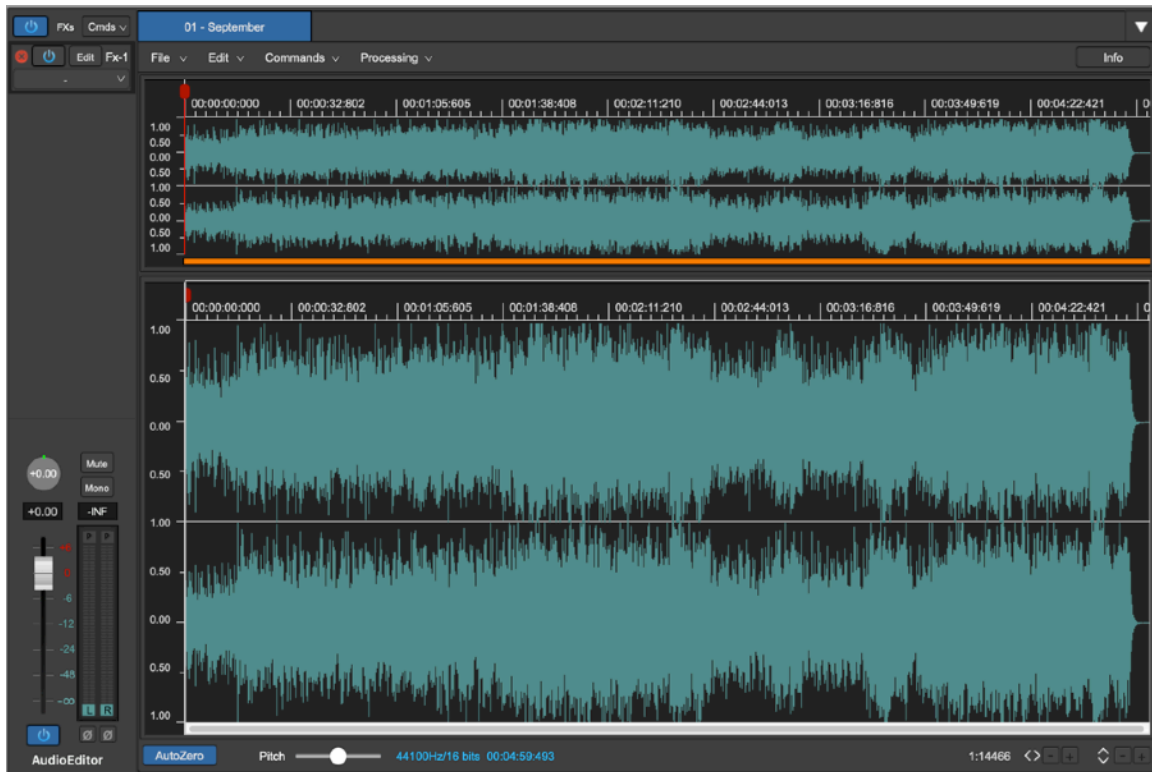
- use the command OpenAudioFile... on the down arrow popup menu control on the top right of the Audio Editor waveform view:



DSP-AudioEditor will open a file browser. Select the audio files you wish to edit.

NOTE: You can select multiple files by \hat{u} +Clicking or \mathbb{S} +Clicking on the file names.

This window interface features these main areas:



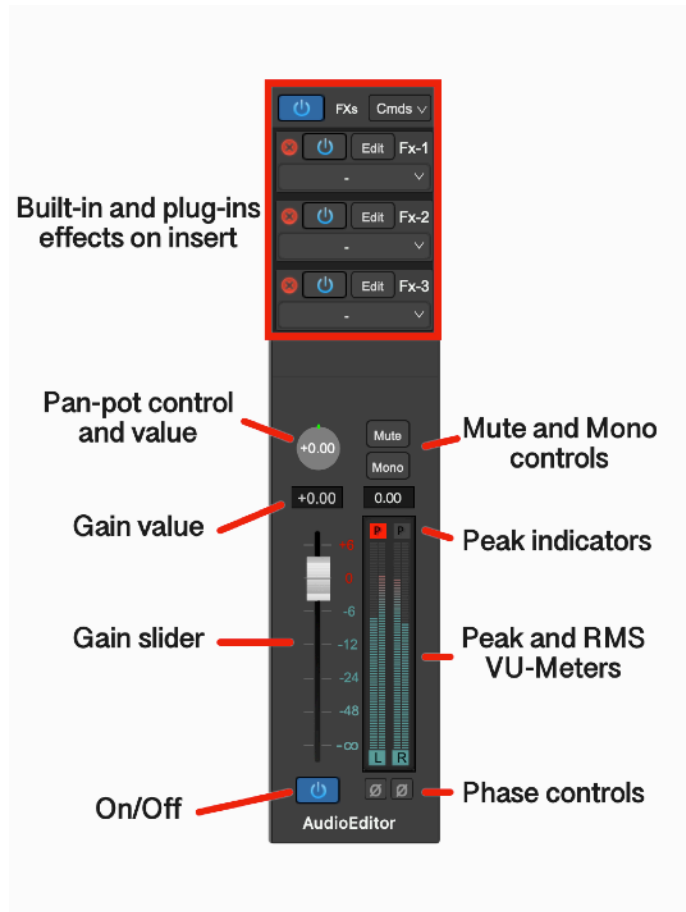
- **The waveform overview:** in the upper part of the audio file editor there is an overview of the audio file under editing. During all editing operations the overview will show you the waveform of the whole audio file, from start to end. The overview zooms in or out automatically if the file changes its length doing changes.
- **The Audio File Editor waveform view area:** in the middle part of the audio file editor there is also the waveform view which, differently from the overview, allows to zoom-in or zoom-out on any region of the audio file. When the waveform zooms, the underneath scroll bar can be used to scroll the view along the x-axis, back and forth in time. The audio file editor waveform area is the main part for the edits. It is possible to set the level of zoom very easily by using the mouse wheel, by using the commands in the menu, by the keyboard, or clicking on the zoom buttons located in the lower right angle of the waveform view area.
- **The Audio File Editor channel strip:** in the left part of the Audio File waveform view, there is a channel strip, with the gain sliders and Vu-Meters section. More about the channel strip later in this manual.

The Channel Strip

On the left part of the Audio File editor waveform view there is a channel strip. Use the channel strip to set the gain, the pan-pot and the other parameters and features - as a serial chain of real-time effects - used by DSP-AudioEditor to play the audio file under editing.

Each audio file under editing has its own channel strip having different settings.

On the lower part of the channel strip there are the Gain sliders and VU Meters:



The Gain sliders, Peak and RMS VU-Meters and other controls

Using the Gain Sliders and VU Meters on the channel strip is very intuitive:

Using the Gain Sliders:

- **Click&Drag:** clicking and dragging the Gain Slider knob will change the gain of the channel according to the dB scale.
- **CTRL + click:** to jump to the default +0dB value.

Using the numerical control above the Gain Slider:

- **Click:** a small dialog will prompt you to enter a new numerical value, expressed in dB (float number).

Peak and RMS Meters:

On the right of the Gain slider there are the Peak and RMS Meters. They show the peak and RMS levels of the left and right signals on DSP-AudioEditor output. The peak meters are innermost respect the RMS meters.

Peak meters show the max level of the audio signal, computed with the precision of the sample. They have a release time such that when the signal decreases, they follow smoothly. As soon as the signal increases, they immediately update the displayed value.

RMS meters show the power of the audio signal. An audio signal can have very high peak levels (even up to the distortion) but very low RMS values. In this case, they sound poor and weak. A good mastering engineer is able to rise the RMS level without causing the distortion. Usually, multi-band compressors plug-ins can help a lot to reach a good balanced result.

In the upper part of the Vu-Meters there is a number that shows the peak value in dB. This number remains fixed to the highest reached value for 3 seconds. Then, it will return to show the current maximum value. In Preferences, it is possible to set a flag to keep this value fixed to the maximum. In this case, to reset it, click on it.

Also there is a red peak level indicator that flashes if the signal reaches or exceeds 0dB. Be careful not to let these indicators flash too often because, even if DSP-AudioEditor, just like analog devices, uses a special clipping algorithm any abuse of the output levels will cause the saturation and distortion in the audio output signal.

The Panpot Rotary Control:

On top of the Gain slider there is a rotary control, the pan-pot, to set the balance between the L&R channels. Possible values are in the range [-1.0..1.0], where -1.0 means that the only the L channel will play, 0 means that the L&R output signals are equally balanced, and +1.0 that only the R channel will play.

It is also possible to click on the numerical value into the rotary control to enter the exact pan-pot value.

ON/OFF button:

It turns On/Off the audio on output from the channel strip.

Mute button:

Use this button to mute/un-mute that channel strip.

Mono button:

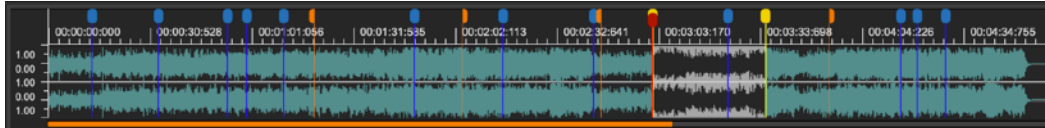
Use this button to mix the L&R channels into a single mono channel, which will play on both L&R channels on the channel strip output. This is very useful to verify the mono compatibility of the output audio signal.

L&R Phase Invert button:

At the bottom of each VU-Meter there is a button with a Ø sign. Turning ON this button will invert the phase of that audio channel. Use this button to check if there is a frequency cancellation when inverting the phase of one channel. These buttons are very useful when used in combination of the Mono button.

The Audio Overview view

The waveform overview is in the uppermost section of the Audio Editor waveform view.



This view is a waveform view without the zoom in/out control.

It is possible to use the waveform overview to make new waveform selection, move the markers or the playback cursor, and set the zoom level for the main waveform view area below.

At the top of the waveform there is the Time Ruler. DSP-AudioEditor uses the same X-Axis unit used for the Counter Display control. As described into the corresponding chapter, to set the X-Axis unit for the time ruler, use the application menu command

Options->X-Axis Units...

Possible choices are

- HH:MM:SS:MSEC (hours:minutes:seconds:milliseconds)
- Frames
- Seconds
- Milliseconds
- AudioCD (CDDA)

On the right side there is the Level Ruler, that shows the level of the signal accordingly to the Y-Axis unit. To set the Y-Axis unit, use the menu command

Options->Y-Axis Unit...

Possible choices are

- Floats, normalized into the range [-1.0..1.0[
- dB (decibel).

Below the waveform display, there is a special **Zoom Controller**, that can be used to set the zoom level of the main waveform view area. This control highlights in orange the region that the main waveform view area displays. By clicking on it and then dragging the mouse, DSP-AudioEditor set the zoom level of the main waveform view scrolling it accordingly.

Using the Zoom Controller

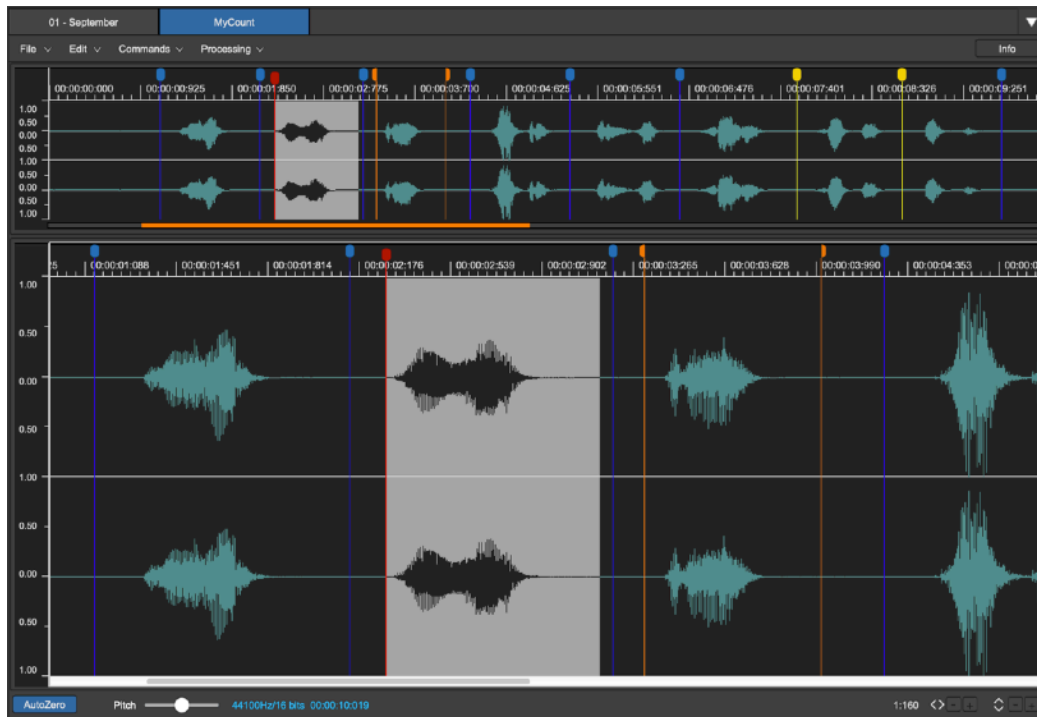
It is easy to use the zoom controller:

- **hover** the mouse over the zoom controller, the mouse pointer changes to a magnifying glass.
- **clicking on it and dragging** the mouse, the control draws an orange region. Then, releasing the mouse button, the main waveform view scrolls

to the position corresponding to the orange rect start, zooming up to the position corresponding to the orange rect end.

- **⌘ + mouse click and dragging**, scrolls the main waveform view to the region pointed by the mouse, without modifying the zoom level.

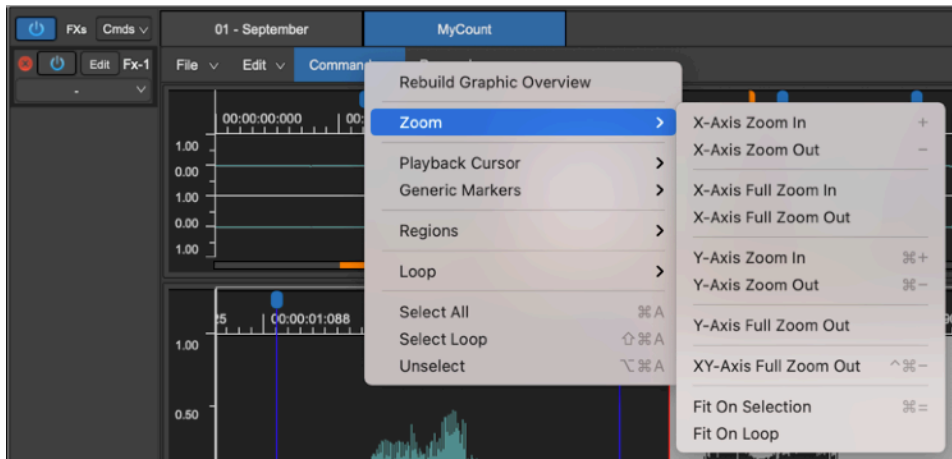
The Main Audio Editor Waveform view



As we have seen for the overview, the main Audio Editor waveform view area shows the waveform of the current audio file. This view however is able to zoom in any part of the waveform, up to one samples per pixel and even more. Both x and y axes have an independent zoom levels. The vertical axis can be zoomed-in without limits, it is even possible to see the quantization noise in the audio file.

To zoom in/out:

- **Use the mouse wheel**: it sets the horizontal X-Axis zoom-in/out ratio. The same can be done moving vertically 2 fingers on the Apple Trackpad, if any. Use **⌘ + mouse wheel** to set the vertical Y-Axis zoom-in/out ratio.
- **Use the +/- buttons**: use the +/- at the bottom right of the waveform view to zoom in/out horizontally on the X-Axis or vertically on the Y-Axis.
- **Use the menu commands**: clicking on the Commands button which is on top of the waveform overview, a popup menu appears, and it is possible to use one of commands on the Zoom sub menu:



- **X-Axis/Y-Axis Zoom In/Out** will increase or decrease the zoom in/out ratio by a factor 2 (it will magnify the waveform by a factor x2 or will zoom-out by a factor x0.5)
- **Full Zoom In/Out:** it is a shortcut to the maximum/minimum possible level.

Both functions are available for the X-Axis X (horizontal), Y-Axis (vertical) or in pair (XY-Axis Full Zoom Out).

There are also some functions to fit the zoomed area to a special region of waveform:

- **Fit On Selection:** to zoom in/out on the current selection. This command is not available if there is not a selection.
- **Fit On Loop:** to zoom in/out on the loop region. This command is not available if the loop is turned OFF.
- **Using the Keyboard shortcuts:** you can use the keyboard as shortcut to the above menu commands.

After zooming the waveform, the scroll bar at the bottom of the main waveform view lets you to scroll the view to another position.

Do not forget to use the Zoom Controller on the upper waveform overview view for immediate zoom in/outs!

NOTE: you can scroll the view also by clicking and dragging the Time Ruler. The effect is the same as using the scroll bar, but often it is smoother and very much more precise.

If it is playing, the Audio Editor waveform view area will scroll following the playback cursor.

In the left area at the bottom of the main waveform view, you can see information such as the sample rate, bit depth and total length of the audio file under editing.

The button AutoZero lets you to toggle On/Off the AutoZero options when placing markers and selection start/end (more about this later in this chapter).

The Info button opens the side panel on the right of the main waveform view, described here below.

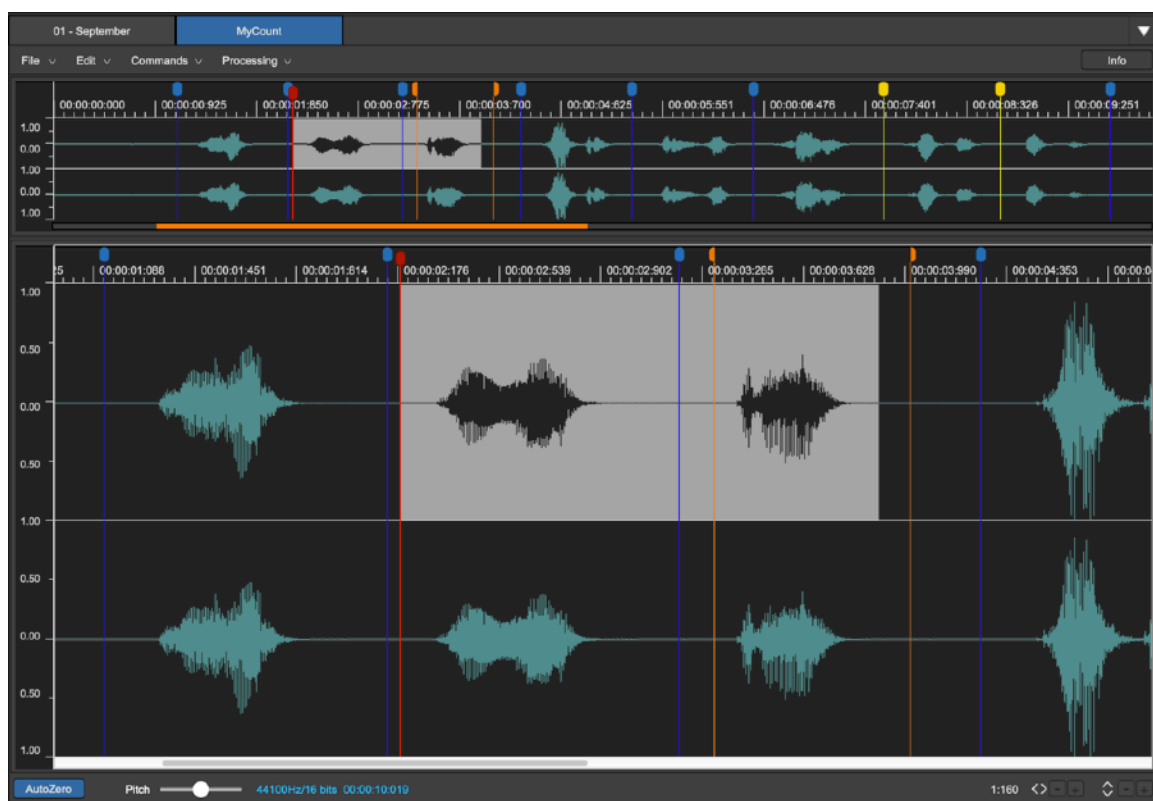
Making a waveform selection

A “selection” is a region of the audio file which we wish to edit. Most of DSP-AudioEditor editing functions work on the selection, if any. Otherwise, they work on the whole audio file.

There are several ways to do a selection: clicking and dragging the mouse, using menu commands or using keyboard key commands.

It is possible to make a selection on the main waveform view area or on the waveform overview. Since both views refer to the same file, a selection done on a waveform view will be visible also on the other waveform view as well. Doing a selection on the main waveform view area, dragging the mouse pointer outside the right or left borders, the view will scroll. The scrolling speed will increase/decrease accordingly to the distance outside the border.

NOTE: If there is already a selection, **⇧+click** will extend the selection up to the mouse pointer. If there isn't already a selection, **⇧+click** creates a new selection between the playback cursor and the mouse pointer.



NOTE: you can also independently select the left or right channel of a stereo audio file. To select only the right channel, click and drag on the topmost part of the waveform of the right channel. To select only the left channel only, click and drag the lower part of the waveform of the left channel. In the case of a selection of only the left or right channel, if applicable, the edit function will process only that channel.

Clicking and dragging the mouse, doing a selection, the Counter Display will show in real time the selection length, start and end points.

NOTE: double-click with the mouse to select the portion of waveform between the two nearest markers. If there are none, all the waveform is selected. A triple mouse click always selects all the waveform.

Clicking on the Commands button which is on top of the waveform overview, It is possible to use the following commands to do selections:

Select All (or ⌘A):

To create or extend the current selection to all the waveform.

Select Loop (or ⇧⌘A)

available only if loop is On, to select the region between start and end loop points.

The audio file Info side panel

Clicking on the Info button which is at the top left of the waveform overview view (or on the rightmost button among the buttons located at the top left of the worksheet, marker with an Info symbol), a side panel will open on the right of the waveform view itself. This side panel has some tab buttons. Clicking on each of them, the view below switches among the Info, Markers, Regions and MIDI views.

The screenshot shows a side panel with four tabs: Info, Markers, Regions, and MIDI. The 'Info' tab is selected. The panel displays the following information:

- Source Audio File Path:** /Volumes/MacBook HD2/Fl...-01/Audio Files/MyCount.aif
- Audio File Name:** MyCount
- Audio File Length:** 00:00:10:019 (Time) / 441856 (Frames)
- File Format:** AIFF
- Sample Rate:** 44100
- Bit/Sample:** 16
- CD-Texts:**
 - Title: MyCount
 - Performer: Steve's family
 - Songwriter: Nicola
 - Composer: Edoardo
 - Arranger: Matteo
 - Message: Zdenka is my life
 - Genre ID: Other (dropdown menu)
 - ISRC: IT0123456789
- ID3 Tags:**
 - Album: (empty field)
 - Genre: Other
 - Year: 2022

The same button used to open the side panel will close it if pressed again.

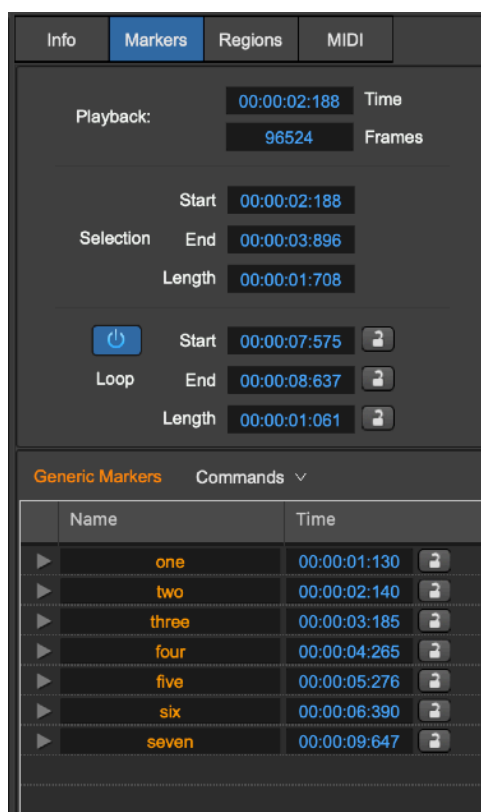
The Info panel is divided into two main parts:

- On top there are read-only info: source audio file path, audio file name, the length expressed in hh:mm:sec:msec and frames units, the audio file format, sample rate and bit depth.
- The lower part shows user editable settings: CD-Texts and ID3 Tags. Click on the editable field to enter the new value, then hit return (or click outside the popup view used to enter the new text) to confirm.

NOTE: please consider that exporting the audio file, not all audio file formats support the CD-Texts and ID3 Tags information.

Working with Markers

Two other tabs on top of the side panel switch to the Markers or Regions views.



There are several types of markers:

- **The Playback Cursor (the RED marker)** - always positioned at the current playback position.
- **Generic Markers (BLUE markers)** - a generic marker is identified by a location and a name.
- **Region markers (ORANGE start and end markers)** - a region is a portion of the audio file which is identified by two markers, one at region start, one at region end, and by a region name.

- **Loop markers (YELLOW start and end markers)** - A Loop is a portion of the audio file to cycle when playing it. It is identified by two markers, one at loop start, one at loop end.

NOTE: DSP-AudioEditor is compliant with the standards when it reads/writes markers information from/to PCM uncompressed audio files (AIFF, WAVE, CAF). These audio files can be exchanged among different applications and software/hardware samplers, maintaining the information about loop and other markers.

NOTE: DSP-AudioEditor gives you an option of **Auto Zero**. When this option is ON, releasing the mouse after moving a marker or setting the selection, DSP-AudioEditor automatically shifts the cursor point to the previous zero-crossing point. A zero-crossing point is the location in which the waveform intersects the zero level changing the sign. This option is very useful to place markers in positions that have the zero level in common.

To activate the Auto Zero option, use the Auto Zero button at the lower left of the main waveform view, or use the menu command **Options->Auto Zero**, or to use the combination of keys **⌘U**. This command toggles the AutoZero ON/OFF.

The Markers view shows the Loop and the Generic Markers, the Regions view shows the Regions markers.

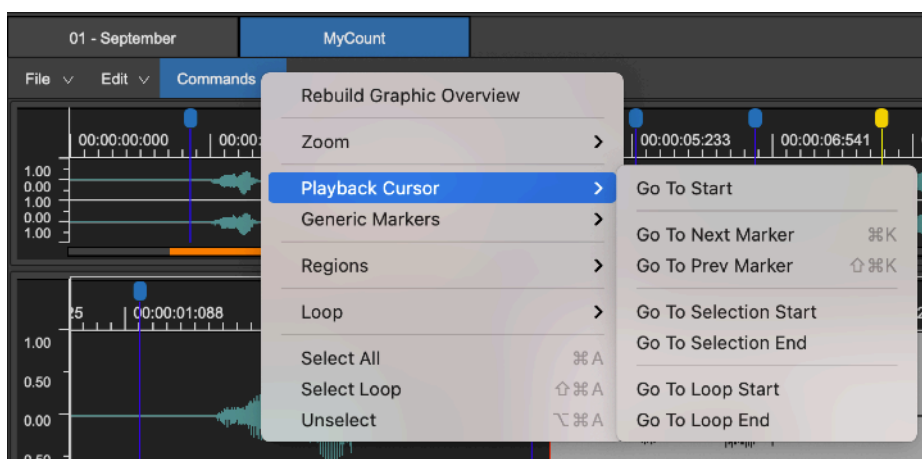
On the Marker view there are two main parts: on top there is a part to show the current values of the playback cursor, of the loop markers (they are visible only if the Loop is ON) and the start and end values of the waveform selection, if any. The bottom parts has a table showing the Generic markers list, and a Command button to work on the Generic markers.

To edit the value of a marker position, click on the numerical field of the value. DSP-AudioEditor will show a popup, where it is possible to type a new value. Use the tab key to move from one field to the next, the enter key to close the popup.

Menu commands to move the playback cursor:

There are several menu commands to move the playback cursor at specific points of the audio file.

Clicking on the Commands button on top of the waveform view, and then on the Playback Cursor submenu, it is possible to:



Go To Start:

to move the playback cursor at audio file start.

Go To Next Marker (or ⌘K):

to move the playback cursor to the position of the next marker.

Go To Prev Marker (or ⇧⌘K):

to move the playback cursor to the position of the previous marker.

Go To Selection Start:

to move the playback cursor to the current audio selection start. This command is not available if there is not an audio selection on the waveform.

Go To Selection End:

to move the playback cursor to the current audio selection end. This command is not available if there is not an audio selection on the waveform.

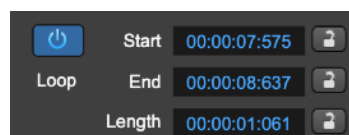
Go To Loop Start:

to move the playback cursor to the Loop start. This command is not available if the Loop is OFF

Go To Loop End:

to move the playback cursor to the Loop end. This command is not available if the Loop is OFF

Among Loop options on the Markers side panel, there are also the buttons to turn On/Off the loop (the yellow start/end loop markers will appear/disappear on the waveform views) and to lock/unlock the loop points and loop length.



If one or both start/end loop markers are locked, the corresponding marker cannot be dragged or moved. And, if the loop length is locked, setting the start loop marker position will set also the end loop marker accordingly. Of course, the reverse is also true.

The Generic markers table

At the bottom of the Markers side panel, there is the Generic markers table:

Generic Markers		Commands ▾
	Name	Time
▶	one	00:00:01:130 🔒
▶	two	00:00:02:140 🔒
▶	three	00:00:03:185 🔒
▶	four	00:00:04:265 🔒
▶	five	00:00:05:276 🔒
▶	six	00:00:06:390 🔒
▶	seven	00:00:09:647 🔒

All fields are editable. Just click on the field to edit it, DSP-AudioEditor will show a popup to enter a new value.

On the right of each generic marker name there is an arrow which turns on red if the corresponding generic marker is the current selected generic marker. If you click on a blue generic marker on the waveform view, that becomes a selected generic marker.

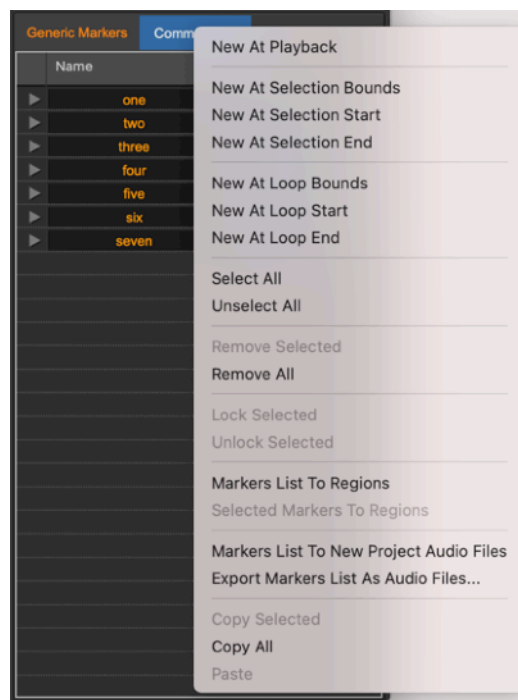
NOTE: it is possible to define a group of generic markers, selecting them at the same time, just keeping the \mathbb{M} or the \hat{u} key pressed while clicking on the arrow on the left part of the generic marker row - or on top of the blue generic marker on the waveform view.

If there are more than one generic marker selected, dragging one of them on the waveform view will also make all the others slide along the time axis.

Generic markers can be locked. In this case, it is not possible to change their position, and it is not possible to move or drag them on the waveform view. If in a group there is a locked generic marker, it is not possible to move any of the others as well.

The Generic markers commands

Clicking on the Commands button on top of the generic markers table, DSP-AudioEditor shows the following menu:



Use these menu commands (menu items are greyed-out if the command not applicable in a certain situation) to:

New At Playback (or \mathbb{M}):

to create a new generic marker at playback cursor position

New At Selection Bounds (or ^⌘M):

If there is a selection, it creates two new generic markers at selection start and end positions. This command is not available if there is not a waveform selection.

New At Selection Start:

if there is a selection, it creates a generic marker at selection start. This command is not available if there is not a waveform selection.

New At Selection End:

if there is a selection, it creates a generic marker at selection end. This command is not available if there is not a waveform selection.

New At Loop Bounds (or ⌘M):

if the Loop is On, it creates two new generic markers at loop start and end. This command is not available if the Loop is OFF.

New At Loop Start:

if the Loop is On, it creates a generic marker at loop start. This command is not available if the Loop is OFF.

New At Loop End:

if the Loop is turned On, it creates a generic marker at loop end. This command is not available if the Loop is OFF.

Select All:

to select all the generic markers. This command also groups them all together.

Unselect All:

to un-select all generic markers. This command is not available if there are not selected markers.

Remove Selected (or ⌘M):

to remove (delete) the selected generic marker(s). This command is not available if there are not selected markers.

Remove All (or ^⌘M):

to remove (delete) all generic markers

Lock Selected:

to lock the selected generic markers. This command is not available if there are not selected markers.

Unlock Selected:

to unlock the selected generic markers. This command is not available if there are not selected markers.

Markers To Regions:

to create new regions from the generic Markers on the list. Each region will extend from one generic marker to the next one on the list, and so on, one after the other.

Selected Markers To Regions:

as above, with the difference that only the selected markers are considered.

Markers List to New Project Audio Files:

to create new audio files from the generic markers list, one for each marker. Each new audio file will extend from one generic marker to the next one on the list.

Export Markers List As Audio Files...:

to export to MacOS file system new audio files from the generic markers list, one for each marker. Each new audio file will extend from one generic marker to the next one on the list. Then, DSP-AudioEditor shows a dialog with the options to export the audio files and to ask a destination. Please refer to the chapter describing the options available to export audio files for the details.

Copy Selected:

to copy the list of selected generic markers into a special clipboard built-in into DSP-AudioEditor. This special clipboard is dedicated only to the generic markers. After copying a list of generic markers into the marker clipboard, it will be possible to paste this list from the clipboard into a new audio file.

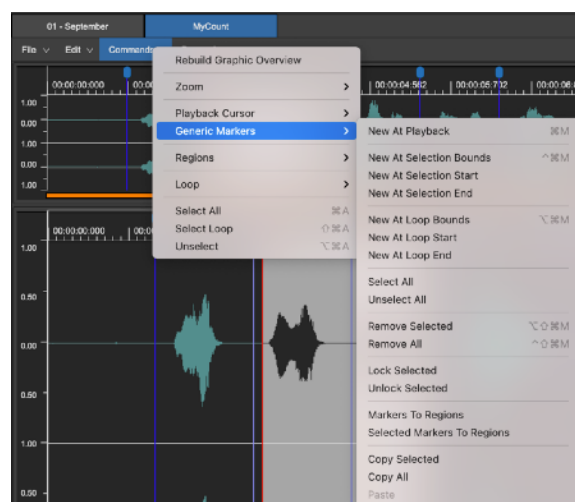
Copy All:

as above, with the difference that this command copies all the generic markers into the marker clipboard.

Paste:

to paste the list of the generic markers which are memorised into the marker clipboard into the current audio file. This command is not available if the marker clipboard is empty.

The same commands are available from the Commands button on top of the waveform view:



Working with Audio Regions

As already introduced, an audio region is a portion of the audio file which is identified by two markers, one at region start, one at region end, and by a region name.

Info	Markers	Regions	MIDI
Regions Commands ▾			
Name	Time		
▼ Intro		⌵ ⌵	
Intro - Start	00:00:01:288		
Intro - End	00:00:02:995		
Intro - Length	00:00:01:706		
▼ Bridge		⌵ ⌵	
Bridge - Start	00:00:04:022		
Bridge - End	00:00:06:286		
Bridge - Length	00:00:02:263		
▼ 2nd bridge		⌵ ⌵	
2nd bridge - Start	00:00:07:348		
2nd bridge - End	00:00:09:159		
2nd bridge - Length	00:00:01:810		
▼ solo		⌵ ⌵	
solo - Start	00:00:03:312		
solo - End	00:00:04:011		
solo - Length	00:00:00:699		

The Audio Regions table

One of the tabs on top of the Info side panel switch to the Audio Regions view. DSP-AudioEditor allows you to add and remove audio regions, edit region name, show/hide the region markers on the waveform view. There are also several commands dedicated to audio regions.

Each row on the table is dedicated to the parameters of an Audio Region :

▼ Intro		⌵ ⌵
Intro - Start	00:00:01:288	
Intro - End	00:00:02:995	
Intro - Length	00:00:01:706	

Click on the arrow on the left side of the audio region row to expose/collapse the details about the audio region start and end positions, and the region length.

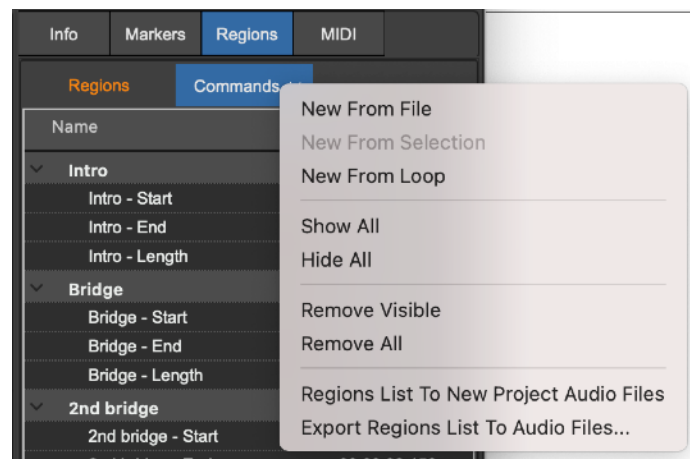
Click on the blue check mark on the right of the audio region row to show/hide the audio region markers (the orange markers) on the waveform view.

The minus (-) button on the right of the audio region row will remove (delete) that audio region.

You can edit the audio region name, start, end and length values just clicking on the corresponding text or value. DSP-AudioEditor will show a popup view to enter the new value.

The Audio Regions commands

Clicking on the Commands button on top of the Audio Regions table, DSP-AudioEditor shows this menu:



New From File (or $\text{⌘} \text{⇧} \text{⌘} \text{U}$):

to create a new audio region starting at audio file start and ending at its end.

New From Selection (or $\text{⌘} \text{⌘} \text{U}$):

to create a new audio region starting at selection start and ending at selection end. This command is not available if there is not a waveform selection.

New From Loop:

to create a new audio region starting at Loop start and ending at Loop end. This command is not available if the Loop is Off.

Show All (or $\text{⌘} \text{⇧} \text{⌘} \text{U}$):

to make all the audio region markers visible on the audio waveform view.

Hide All (or $\text{⌘} \text{⇧} \text{⌘} \text{U}$):

to hide all the audio region markers on the audio waveform view.

Remove Visible:

to remove (delete) all visible audio regions.

Remove All:

to remove (delete) all audio regions.

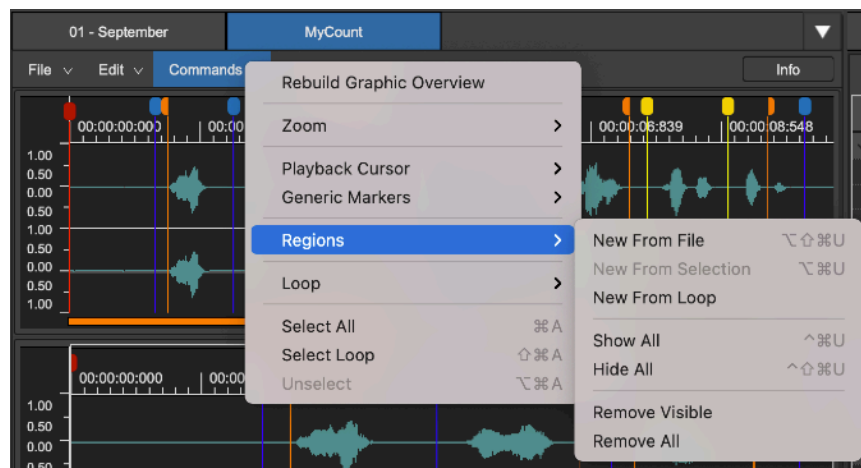
Region List to New Project Audio Files:

to create new audio files from the audio regions list, one for each audio region.

Export Region List to Audio Files....:

to export to MacOS file system a list of new audio files from the audio regions list, one for each audio region.

The same commands are available from the Commands button on top of the waveform view:



Working with the Loop

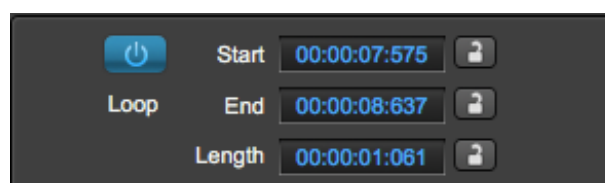
When it play an audio file, DSP-AudioEditor uses two different concepts for cycling on a certain region of an audio file: the Loop and the Cycle.

The Cycle Mode and the Loop

The **Loop** is a region of the audio file to cycle when playing it. It is identified by two markers, one is the loop start, the other one is the loop end (the yellow markers on the waveform view). These two markers are saved into file, and into the Project.

Also the **Cycle** is a portion of the audio file to cycle when playing it, BUT it is not identified by any marker. It is simply a region which DSP-AudioEditor will play again and again without any interruption.

The Loop can be ON or OFF accordingly to the status of the On/Off button on the audio file Info side panel, among Markers settings:



The Cycle can be ON or OFF accordingly to the state of the Cycle button on the Transport bar:



Q: When DSP-AudioEditor plays cycling on the Cycle instead of cycling on the Loop?

A: let's start from **the rule** which DSP-AudioEditor follows to play:

- 1) **If there an audio selection**, DSP-AudioEditor plays that audio selection ONLY, starting at the start of the audio selection and ending at selection end.

Then, if the Cycle is ON, DSP-AudioEditor plays cycling on that audio selection. If the Cycle is OFF, it stops as soon as it reaches the end of the selection. This will happen no matter if the Loop is On or Off.

- 2) **If there is not an audio selection**, DSP-AudioEditor plays the audio file starting at the beginning of the audio file. Then:

- If the Loop is ON, DSP-AudioEditor plays cycling on the Loop (yellow markers on the waveform view).

- If the Loop is OFF, DSP-AudioEditor plays the audio file up to the end, then it will cycle starting again at the beginning of the audio file ONLY if the Cycle button is ON.

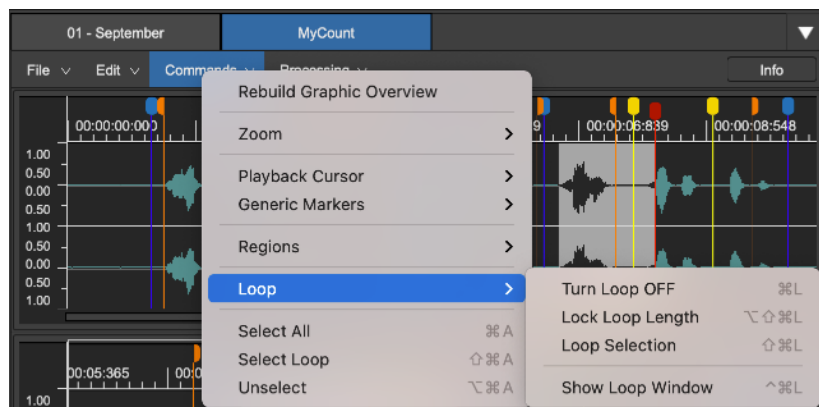
- If both Loop and Cycle button are OFF, DSP-AudioEditor will stop to play as soon as it reaches the end of the audio file.

NOTE: if there is an audio selection on the waveform view, DSP-AudioEditor plays that selection only, cycling on it if and only if the Cycle button is ON.

NOTE: and, if there is an audio selection on the waveform view, when playing that selection, DSP-AudioEditor does not consider the Loop (yellow markers) settings, even if the Loop is ON.

The Loop commands

Clicking on the Commands button on top of the waveform view, DSP-AudioEditor shows a menu which includes the following commands for Looping:



Turn Loop ON/OFF (or ⌘L):

to toggle the Loop ON and OFF. If the Loop has been turned ON, the two Loop markers (the yellow markers), will appear on the waveform view. Otherwise, DSP-AudioEditor will hide them.

Lock Loop Length (or ⌘⇧L)

to lock/unlock the length of the Loop. If the Loop length is locked, moving any one of the two Loop markers, the other one will move accordingly. This option is useful in the case that, once you have found a good Loop, you wish to shift the Loop without changing its length. This command is not available if the Loop is OFF.

Loop Selection (or ⌘⇧L):

to place the Loop markers at the current selection start and end. (This command is not available if there is not a selection on the waveform view). If the Loop is turned Off, it will automatically be turned On.

Show/Hide Loop Window (or ⌘L):

to show or hide the Loop floating window tool. More about this here below.

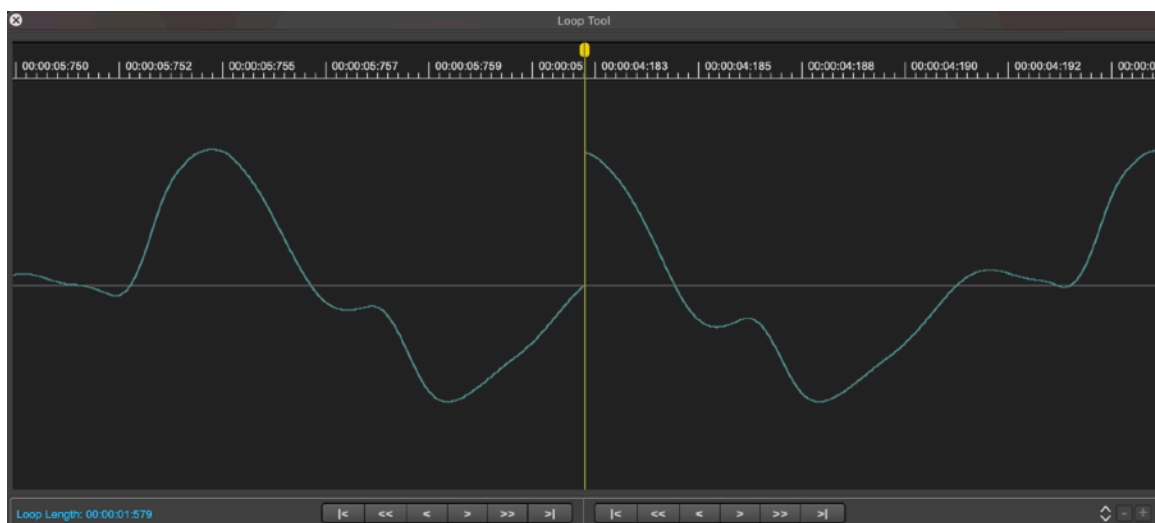
The Loop Window

DSP-AudioEditor gives you a special tool for setting Loop points: the Loop Window and a very special X-Fade Looping tool. In this chapter the Loop Window will be described. To know about the X-Fade Looping tool, please refer to the chapter describing the X-Fade Looping processing function of DSP-AudioEditor.

This tool allows you to monitor the waveform in correspondence of the Loop points.

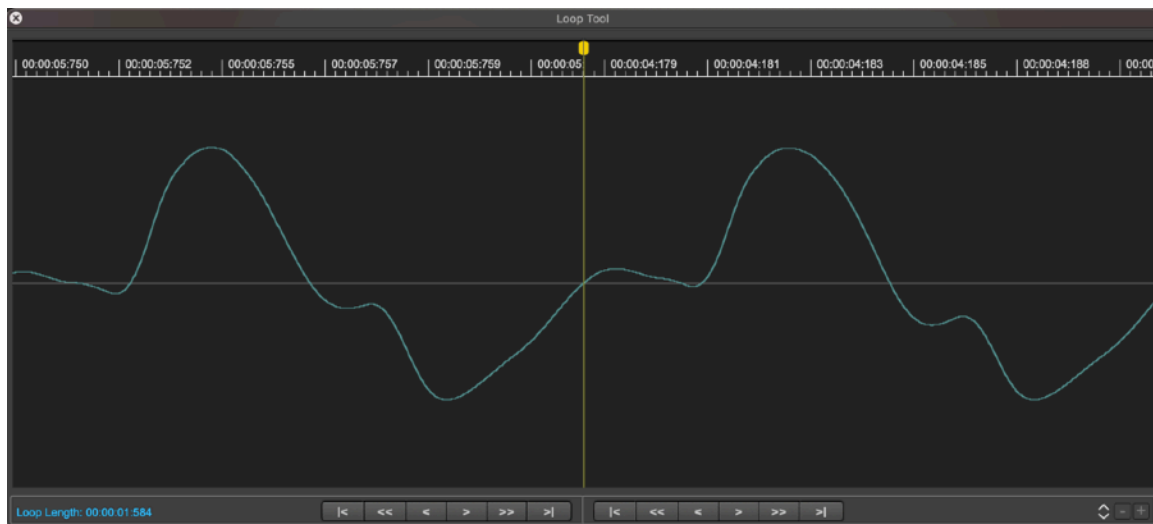
NOTE: the window is a floating window. It means that it is always visible on top of the Editor window. At the same time, it is also possible to work on the waveform Editor view.

Any operation done on the waveform Editor view, like dragging a Loop marker, will result in a real time change in the Loop window, and vice versa.



This window consists of two parts, the left and right parts, divided by a yellow marker that corresponds to the loop point, where DSP-AudioEditor jumps back cycling: the left side of the view shows the final part of the waveform just before the Loop End point, the right side of the view shows the part of waveform which is immediately after the Loop Start point.

When playing the Loop, DSP-AudioEditor plays the samples up to the Loop End point, then it jumps and plays without any interruption from the sample at the Loop Start point. The Loop Window shows you what DSP-AudioEditor plays when it reaches the Loop point, showing the visual continuity of the waveform at loop. If the two points of start and end loop don't coincide or don't grant a continuity to the waveform, as for example in the first above figure, it is clear that a very unpleasant click will be heard at loop point, when DSP-AudioEditor will cycle jumping from the loop end to the loop start.



On the other hand, if the loop points are placed in such way to guarantee a continuity to the waveform, as shown in this second figure, we may expect that there won't be any audible click playing the loop, when DSP-AudioEditor jumps from the loop end to the loop start.

Unfortunately the visual continuity is not always sufficient to guarantee that the Loop will play without clicking, but surely the Loop Window is an essential tool to set the loop points at very good positions.

After doing this, it is also possible to use the X-Fade Looping algorithm of DSP-AudioEditor to produce a Loop which plays absolutely perfect. Of course, a certain experience is necessary to find the "perfect" Loop, especially for audio files which are harmonically complex, as in the case of samples of orchestral strings or of an acoustic piano.

The Loop Window has its own vertical Y-Axes zoom level. To do vertical zoom in/out, use the two -/+ buttons on the lower right part of the Loop Tool window.



There are also two custom transport bars to move the start/end points. One for setting the loop end, below the left view, one for setting the loop start, below the right view.

Respectively, from left to right, these buttons allow you:

- to Move the Loop point on the previous zero crossing point. A zero-crossing point is the location in which the waveform intersects the zero level changing the sign. This option is very useful to place markers in positions that have the zero level in common, if the the slope of the curve is the same at the end and loop positions. The waveform of a perfect loop has always the same level and same slopes at loop points, and the point of zero crossing can be very useful for having a common reference point for placing both start and end loop points.
- To quickly move the Loop point backward.
- To slowly move the Loop point backward.
- To slowly move the Loop point forward.
- To quickly move the Loop point forward.
- To move the Loop point to the next point of Zero Crossing.

NOTE: you are able to drag forward or backward the Time Ruler of every Loop view. The respective Loop point will move accordingly.

NOTE: to place the loop points in the right places is only the first step to have a file playing perfect when jumping at loop. After placing the loop markers, use the X-Fade looping algorithm (see the processing function in the following chapters) to smooth the waveform at loop. Surely, the result will surprise you.

Editing an Audio File

This chapter describes in deep all the editing functions offered by DSP-AudioEditor and how to use them.

Audio File Formats

DSP-AudioEditor loads audio file previously saved using almost all known formats, starting from the PCM uncompressed formats, like AIFF, WAVE, CAF having 8, 16, 24, 32 and 32 float bits/sample, and any sample rate, to compressed audio file formats as mp3, MPEG-4, mp2, ac3, aifc, 3gp, mpa, mp4, 3g2, m4a, AAC...

DSP-AudioEditor takes care to do all necessary conversion loading an audio file, and this process is totally invisible to you.

As a special option, DSP-AudioEditor is able to import an audio file as **RAW data**. It means that DSP-AudioEditor will read the file, loading it sample by sample without looking for any audio file header, as it usually does. It is useful to recover corrupted PCM files where the header is no longer valid. The RAW data import DOES NOT work at all the formats which are not PCM-based, ie if the source file is using any technique to encode or compress. See the following paragraph to know how to import an audio file as RAW data.

When DSP-AudioEditor needs to edit an audio file, it imports the source audio file into a temporary area, translating it to the internal uncompressed CAF 24 bits audio file PCM format. This is because the CAF guarantees the maximum audio fidelity and supports to audio long files. This means that you can work with audio files without any reasonable restriction on its length, as it happens if using the standard AIFF and WAVE formats.

In fact, AIFF and WAVE formats are limited to 2.1 GB for the single audio file and which is, working at 24bits/96kHz, a little bit more than 1 hour only. Instead, CAF audio files can be as long as needed, so you can work with even hours and hours of music at 24bits/96kHz without any problem.

Playing an audio file, DSP-AudioEditor plays always in sync with the current audio device. It means that its internal audio engine works at the same rate of the current audio device. If the sample rate of the audio file is not equal to that of the current audio device, DSP-AudioEditor applies a real time sample rate conversion (SRC) to play the audio file at the correct pitch.

NOTE: the sample rate of the current audio device and sample rate of the current audio file should have always the same value, otherwise DSP-AudioEditor, playing that audio file, needs to process the audio stream with its SRC algorithm and you do not hear EXACTLY what you are going to save on the audio file after your edits.

So, it would be always important to set the sample rate of the current audio device to the same value of the current audio file under editing OR to resample the audio file under editing to the sample rate of the current audio device using the off-line SRC algorithm of DSP-AudioEditor (it is a very high quality linear phase SRC) before to do your edits with it.

Undo/Redo

DSP-AudioEditor always allows to undo changes, using the **Undo command (or ⌘Z)**. The **Redo command (or ⇧⌘Z)** command does the opposite, restores again to the edited version exactly as it was just before to do Undo.

The Undo/Redo engine of DSP-AudioEditor allows an unlimited number of Undos/Redos: DSP-AudioEditor keeps track of all edits done since it opened the file – until it saves the audio file.

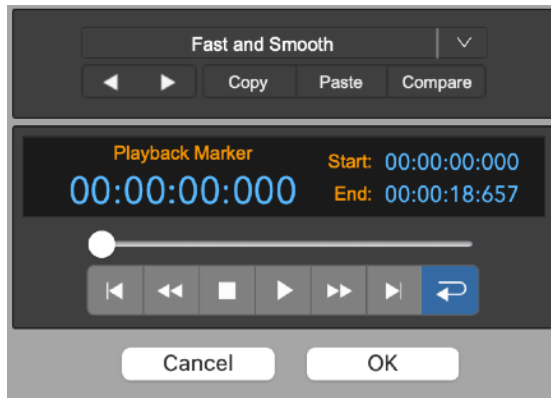
Please beware that saving an audio file also deletes the Undo/Redo history. The same happens when quitting the application.

NOTE: each file under editing has its own Undo/Redo history. Doing some edits on one file, and then switching to a different audio file and doing some edits on this new file, DSP-AudioEditor starts to save a new Undo/Redo history. Then, switching back to the previous audio file, its own Undo/Redo history is safe and it is possible to Undo/Redo last edits done on that file.

Introducing the Real-Time Previewer Controller

For previewing the processed audio, DSP-AudioEditor allows to hear the result of the edit before deciding whether to apply the changes or not. In this way it is possible to set the parameters of the algorithm in real time while hearing the result.

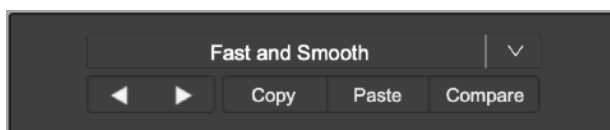
DSP-AudioEditor uses this Real-Time Previewer Controller in several occasions. Usually it is located on the right side of the dialog used to set the parameters which are specific to a certain edit function, and appears like this:



This controller consists of:

The Preset programmer:

On top there is the preset programmer, that manages the preset of the algorithm.

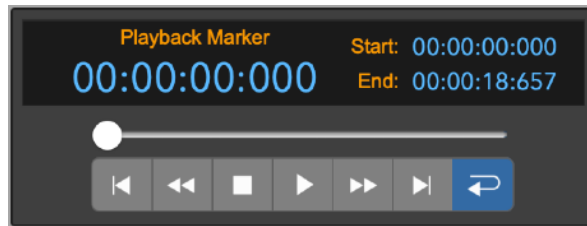


If the algorithm allows to store one or more preset, on top there is an active popup menu which allows to select one of the available preset.

This section allows to set, save and load a preset. A preset is a collection of all the parameters which are necessary to control the algorithm.

The preview transport bar:

On the bottom part there is the Transport bar, to control the playback of the audio file.



Transport buttons and the display are very similar to the transport bar used to control the playback of an audio file under editing. Please refer to the relevant character for the description of the use of each button and of the display.

The slider allows to move the playback cursor back and forth in time, along the audio file.

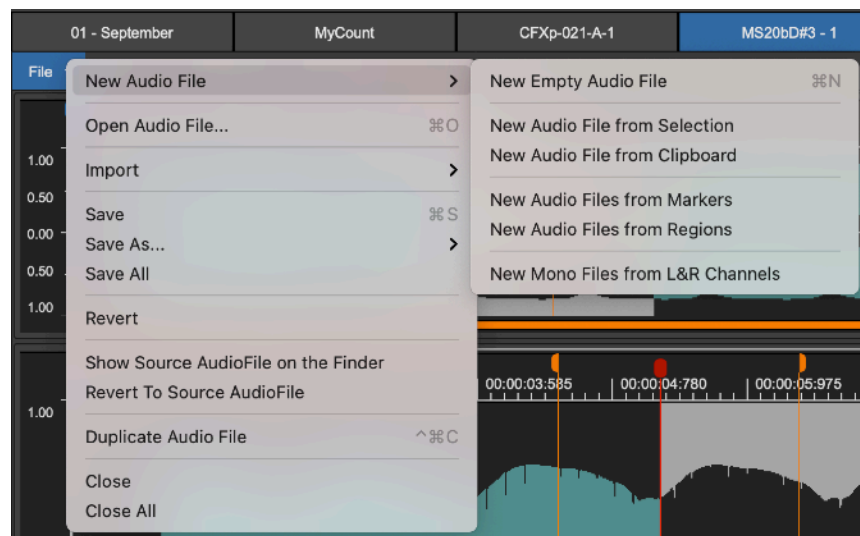
The File Menu Commands

Clicking on the File button on top of the waveform view, DSP-AudioEditor shows a menu with the commands for managing files.

New Audio File

New Empty Audio File (or ⌘N):

to create a new temporary audio file adding it to the Document Manager,



then showing it on a new tab of the waveform Editor view.

New Audio File from Selection:

as above, with the difference that the new audio file is not empty but contains the samples of the source audio file from the selection start to the selection end. This command is not available if there is not a waveform selection.

New Audio File from Clipboard:

to create a new audio file filling it with contents of the current audio clipboard, then adding this new audio file to the Document Manager, and showing it on a new tab of the waveform Editor view. This command is not available if the audio clipboard is empty.

New Audio Files from Markers:

to create new audio files from the list of the generic makers, adding the new audio files to the to the Document Manager. Each new audio file extends from one generic marker to the next one on the list. This command is not available if there are not generic markers.

New Audio Files from Regions:

to create new audio files from the list of the audio regions, adding the new audio files to the Document Manager. This command is not available if there are not audio regions.

New Audio Files from L&R Channels:

to create two new mono audio files, one from the left and one from the right channels of the current stereo audio file, then adding the new audio files to the Document Manager. This command is not available if the audio file is not stereo.

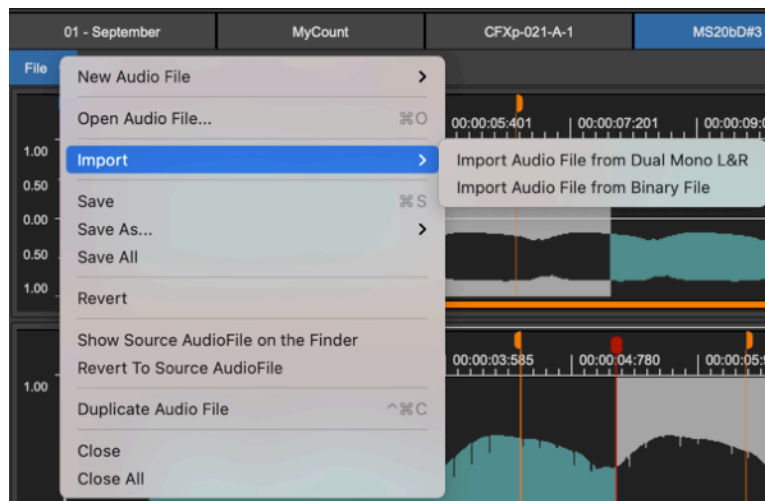
Open Audio File

- **Open ... (or ^⌘O):** to load an audio file from the MacOS file system. DSP-AudioEditor shows a file browser to select one or more audio files to open, then adds them to the worksheet, also showing them on the waveform Editor view.

NOTE: you can select more than one audio file at the same time.

NOTE: DSP-AudioEditor allows to select any audio file in one of the possible supported format (.aif, .wav, .mp3, .m4a, .caf,), then - if necessary - it takes care to translate the audio file to a un-compressed PCM audio file, starting to work with it with the maximum audio quality. In the case in which it is not possible to select an audio file in file browser because it is greyed-out, it would be necessary to check if that the file name ends with the proper extension (.wav, .mp3, ...).

Import Audio File...



Import Audio File from Dual Mono L&R:

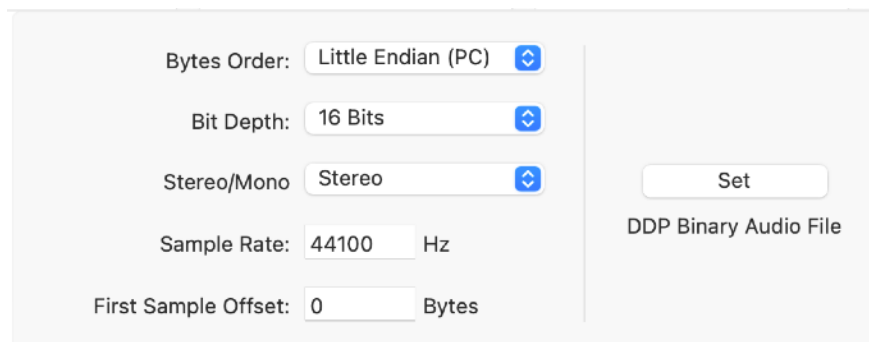
to load a stereo audio file selecting two mono audio files, one for the Left and one for the Right channels (Dual Mono). This command is useful if you need to edit in DSP-AudioEditor a stereo audio file generated by an external application which uses to save a stereo audio file as two -L and -R mono audio files.

Import Audio File from Binary File:

DSP-AudioEditor is able to import an audio file building it reading a RAW binary file. It is necessary to enter some parameters used by DSP-AudioEditor to translate the binary file to an audio stream to save into a new audio file.

DSP-AudioEditor will show a file browser. In the bottom part of this file browser it is possible to set the parameters used to translate the binary audio file to a playable audio file:

Bytes Order: when the below Bit Depth parameter is greater than 8 bits/sample, each sample will use more than one byte. These bytes can be ordered by following the “Big Endian” standard, generally adopted by the Motorola microprocessors and PowerPC-based Macs, or “Little Endian” standard, generally adopted by Intel or Apple Silicon-based Macs and MSDOS/Windows PCs.

The image shows a dialog box for configuring audio parameters. It contains five rows of settings, each with a label, a value field, and a unit or dropdown indicator. The settings are: 'Bytes Order' set to 'Little Endian (PC)' with a dropdown arrow; 'Bit Depth' set to '16 Bits' with a dropdown arrow; 'Stereo/Mono' set to 'Stereo' with a dropdown arrow; 'Sample Rate' set to '44100' with 'Hz' as the unit; and 'First Sample Offset' set to '0' with 'Bytes' as the unit. To the right of these settings is a 'Set' button and the text 'DDP Binary Audio File'.

Bit Depth: to set if DSP-AudioEditor must read 8, 16, 24, 32 or 32 (floats) bits for each sample.

Stereo/Mono: to set the number of interleaved audio channels. For stereo audio files, DSP-AudioEditor will read one sample to fill the left channel, then one sample to fill the right channel, and again.

Sample Rate: to set the sample rate that will be adopted for playing the file.

First Sample Offset: to set the offset, expressed in bytes, from the beginning of the file, that will be used by DSP-AudioEditor before to start to read samples. This value is very important to skip the audio file header correctly to realign the audio sample reading position.

Click on the **Set button** to set the values to the default values to use to read a binary audio file produced accordingly to the DDP standard.

Save

Save (or ⌘S):

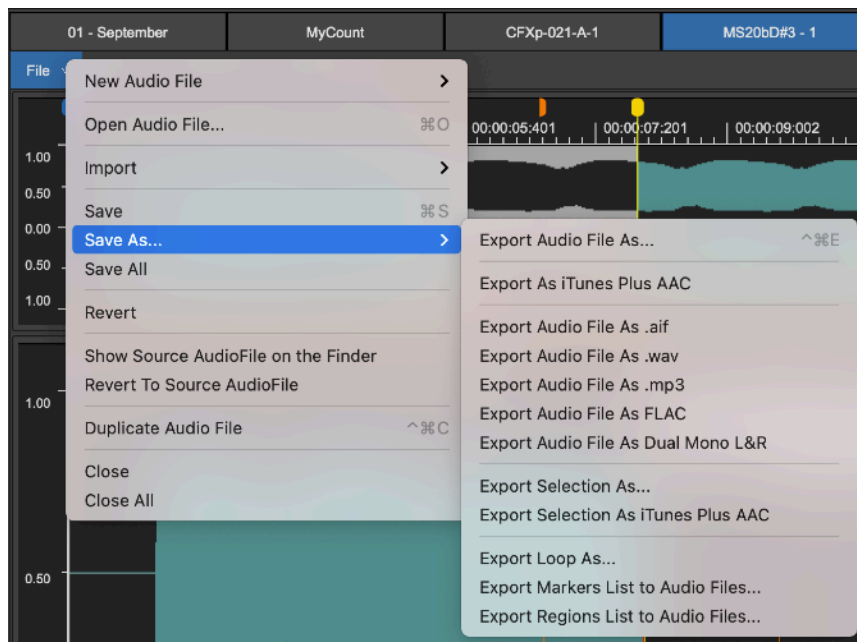
to save the current changes done on the current audio file under editing.

NOTE: DSP-AudioEditor will always ask to save your edits - if any - when it quits.

NOTE: after saving changes, DSP-AudioEditor will reset the Undo/Redo history.

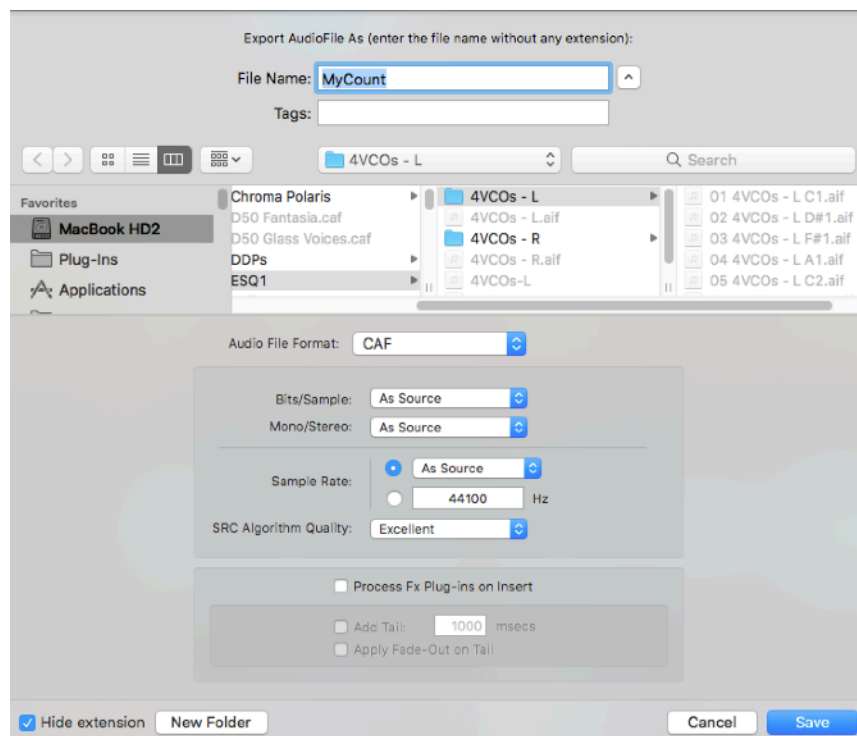
Save As...

The sub-menu contains several commands to export the current audio file converting it using several different audio formats for the destination audio file.



Export Audio File As... (or ⌘E):

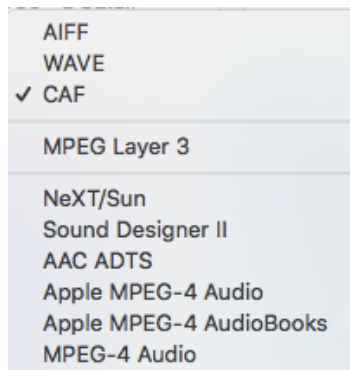
to export the current audio file to the MacOS file system. DSP-AudioEditor will show a file browser to type the destination file name, and with a custom



view to set the audio file format to use exporting samples to the destination file.

The settings on this custom view are:

- **Audio File Format:** to set among the standard formats which are available on the list:



According to the chosen audio file format, the view which is on the bottom part of the the file browser changes.

The uncompressed PCM formats (AIFF, WAVE, CAF) need of these additional parameters:

- **Bits/Sample:** DSP-AudioEditor can save 8, 16, 24, 32 or 32 (floats) bits for each sample. Set "As source" to keep the bit depth of the source audio file.
- **Stereo/Mono:** to set the number of interleaved channels. Set "As source" to keep the number of channels of the source audio file.
- **Sample Rate:** to set the sample rate of the destination audio file. Set "As source" to use the same sample rate value of the source audio file.
- **SRC Algorithm Quality:** if the sample rate of the destination audio file is not the same of the source audio file, DSP-AudioEditor will use its internal Sample Rate Converter (SRC) to write the destination audio file. Four levels of quality of the advanced liner phase algorithm of DSP-AudioEditor are available, ranging from Fast to Excellent.
- **Process Fx Plug-in on insert:** if ON, DSP-AudioEditor will render the source audio file using the FX chain on insert on the audio file channel strip.
- **Add Tail:** if the option to process FX is ON, it is possible to add a tail to the destination audio file, setting also how much this tail must be long. This is very useful if the FX chain includes an effect like a reverb or space simulator which produces an additional tail after stopping the source audio file at its end. Turning ON the option "Add Tail", DSP-AudioEditor will continue to process the audio after reaching the audio file end, capturing the tail of the effect or of the chain of effects.
- **Apply Fade-Out on Tail:** If the options to process FX and to Add Tail are ON, it is possible to apply a fade-out curve to the tail, avoiding an abrupt end at the end of the destination file. Turning ON "Apply Fade-Out on Tail", DSP-AudioEditor decreases the output gain while processing the tail, resulting in a smooth end.

Export As iTunes Plus AAC:

it is a shortcut to “Export Audio File As...” using the iTunes Plus AAC standard audio file format for the destination audio file. In this case, no additional parameters are required.

Export Audio File As .aif:

it is a shortcut to “Export Audio File As...” , setting AIFF as audio file format.

Export Audio File As .wav:

it is a shortcut to “Export Audio File As...” setting WAVE as audio file format.

Export Audio File As .mp3:

it is a shortcut to “Export Audio File As...” setting MPEG Layer 3 (mp3) as audio file format.

Export Audio File As FLAC:

it is a shortcut to “Export Audio File As...” setting FLAC as audio file format.

Export Audio File As Dual Mono:

given a stereo audio file, saves two mono audio files, one for the Left and one for the Right channels (Dual Mono), adding a “-L” to the file name of the first one, a “-R” to the file name of the second one.

Export Selection As...:

calls “Export Audio File As...”, exporting only the region of the selected audio file waveform. This command is not available if there is not an audio selection on the waveform view.

Export Selection As iTunes Plus AAC:

calls “Export Selection As...”, but using the iTunes Plus AAC standard audio file format for the destination audio file. This command is not available if there is not an audio selection on the waveform view.

Export Loop As...:

calls “Export Audio File As...”, exporting only the region of the Loop. This command is not available if the Loop is OFF.

Export Markers List to Audio Files:

to export audio files from the list of the generic makers. Destination audio files will extend from one generic marker to the next one on the list. This command is not available if there is not any generic marker.

Export Regions List to Audio Files:

to export audio files from the list of the audio regions. This command is not available if there is not any audio region.

Save All

Save All:

to save all the audio files that have been changed, one after the other.

Revert

Revert:

to revert changes. DSP-AudioEditor will reload the last saved audio file.

Duplicate Current Audio File

Duplicate Current Audio File... (or ^⌘C):

to make a copy of the current audio file, adding this copy to the worksheet, and showing it on a new tab of the waveform Editor view.

Close

Close:

to close the current audio file under editing. If the file has been changed, DSP-AudioEditor will ask to save changes, to discard or to cancel the operation.

NOTE: to click on the left part of the tab on top of the waveform editor view of a certain audio file is equivalent to use the Close menu command.

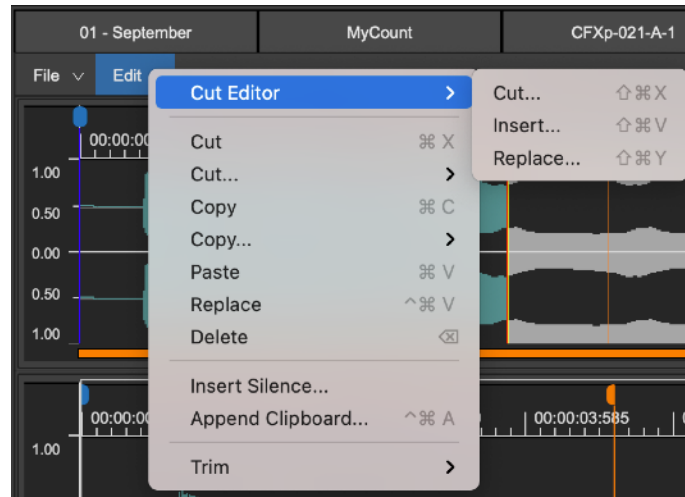
Close All

Close All:

to close all the audio files under editing. For each audio file, if it has been changed, DSP-AudioEditor will ask to save changes, to discard or to cancel the operation.

The Edit Menu Commands

Clicking on the Edit button on top of the waveform view, DSP-AudioEditor shows a menu with the commands for doing cut, copy&paste and similar edits on the audio file:

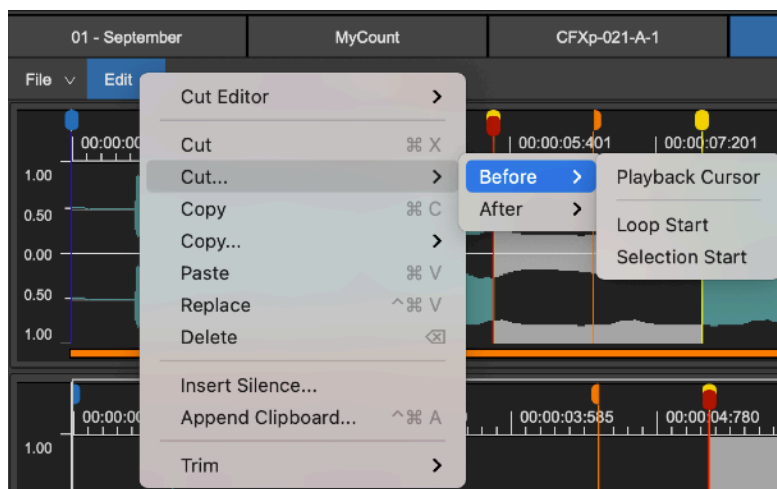


Cut Editor...

The Cut Editor Tool of DSP-AudioEditor is a very unique graphical tool to make cut&paste of audio regions without any problem caused by abrupt changes of the level and slope of waveforms at cut or paste points.

Please refer to the chapter dedicated to the Cut Editor Tool to know more about it and how to work with it.

Cut...



Cut (or ⌘X):

to cut the selected region of the audio file. DSP-AudioEditor will also copy that region into the built-in audio clipboard.

Cut Before:

a shortcut to cut a region of the audio file from the beginning of the audio file up to a certain point. DSP-AudioEditor allows the following commands:

Cut Before Playback Cursor:

to cut the region of the audio file before the playback cursor. This command is not available if the playback cursor is at the beginning of the audio file.

Cut Before Loop Start:

to cut the region of the audio file before the Loop Start. This command is not available if the loop is OFF or if the Loop starts at the beginning of the audio file.

Cut Before Selection Start:

to cut the region of the audio file before the of the audio selection start. This command is not available if there is not a selection or if the selected audio file waveform starts at the beginning of the audio file.

Cut After:

similar to the command above, but to cut a region of the audio file from a certain point up to the audio file end. DSP-AudioEditor allows the following commands:

Cut After Playback Cursor:

to cut the region of the audio file after the playback cursor. This command is not available if the playback cursor is at the end of the audio file.

Cut After Loop End:

to cut the region of the audio file after the Loop End. This command is not available if the loop is OFF or if the Loop ends at the end of the audio file.

Cut After Selection End:

to cut the region of the audio file after the the audio selection end. This command is not available if there is not a selection or if the selected audio file waveform ends at the end of the audio file.

NOTE: on DSP-AudioEditor Preferences, there is an option to add a certain number of frames before/after the cut point when applying the Cut Before, Cut After and Trim (more about the Trim command later in this chapter) commands. This option is very useful to keep an additional small portion of waveform before or after the portion of the audio file to keep. Some softwares or hardware samplers need of some samples after the end loop point to avoid to click at loop, due their internal interpolation algorithm when transposing the sample along the keyboard.

Copy...

Copy (or ⌘C):

to copy a portion or all the audio file into the built-in audio clipboard. If there is a selection on the waveform view, DSP-AudioEditor will copy only the selected region. If there is not a selection, DSP-AudioEditor will copy all the audio file into its audio clipboard.

NOTE: selecting only one channel of a stereo audio file, then doing copy, DSP-AudioEditor copies only that selected channel into the audio clipboard.

Copy Left into Right Channel:

to copy the left audio channel of the stereo audio file into its right channel, overwriting it. This command is not available if the audio file is a mono audio file.

Copy Right into Left Channel:

to copy the right audio channel of the stereo audio file into its left channel, overwriting it. This command is not available if the audio file is a mono audio file.

Paste (Insert) and Replace

Paste (or ⌘V):

if there is a selection on the waveform view, this command replaces the current selection with the current audio clipboard. If there is not a selection on the waveform view, this command inserts the current audio clipboard at the position of the playback cursor. In both cases, replacing or inserting, DSP-AudioEditor places all the audio file clipboard starting at the insertion point (this insertion point is the selection start when replacing, the playback cursor when inserting). DSP-AudioEditor also changes the audio file length accordingly. This command is not available if the audio clipboard is empty.

Replace (or ⌘V):

if there is a selection on the waveform view, this command replaces the current selection with the current audio clipboard. If there is not a selection on the waveform view, this command replaces the current audio file, starting at the position of the playback cursor, with the audio clipboard. This command is not available if the audio clipboard is empty.

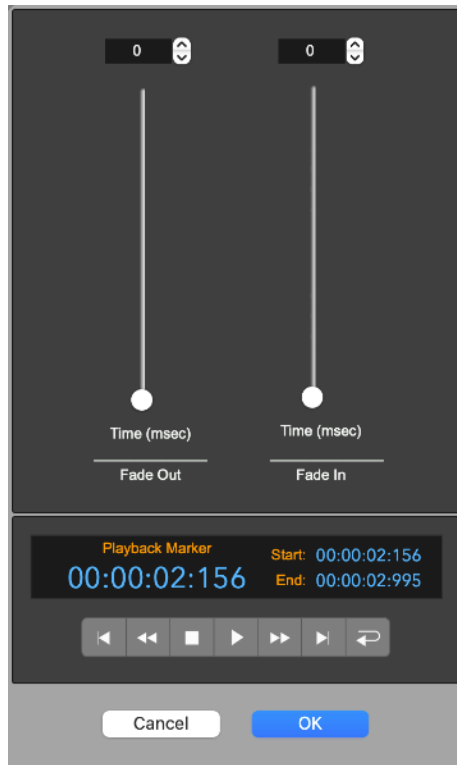
Delete...

Delete (or backspace):

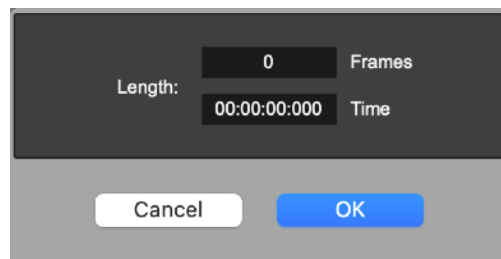
to clear (silencing, as with an eraser tool) the selected waveform or all the audio file if there is not a selection.

DSP-AudioEditor shows an additional view to set the fade-out and fade-in times to avoid abrupt changes. There is also a transport bar to preview the result of deleting.

NOTE: this is a simplified version of the processing function Fade Out<->In, which is very much more powerful. Please refer to the part of this chapter describing the Fade Tool to know more about it.



Insert Silence...



Insert Silence...:

to insert a region of silence (no audio) at playback position.

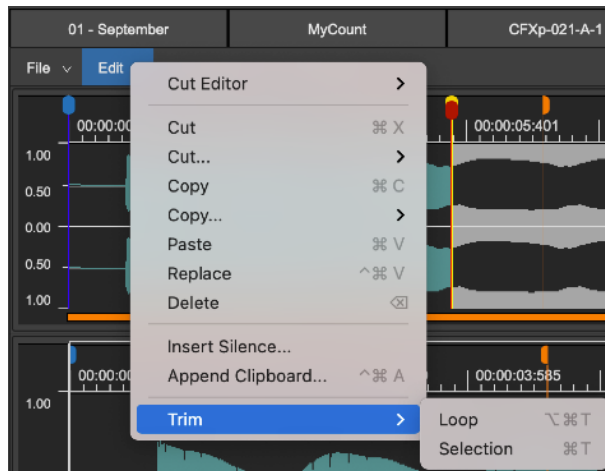
Then, DSP-AudioEditor shows the above dialog to set length of the region to insert, in frames and time units.

Append Clipboard...

Append Clipboard...:

to paste the current audio clipboard at the end of the audio file. This command is not available if the clipboard is empty.

Trim...



to cut the regions outside a certain region, that is to cut the two regions which are before and after the region to trim. It is the same as doing two cuts, one before the region start, one after the region end.

There are two options:

Trim Loop (or ⌘T):

to cut the regions outside the Loop. If the Loop is OFF, this command is not available.

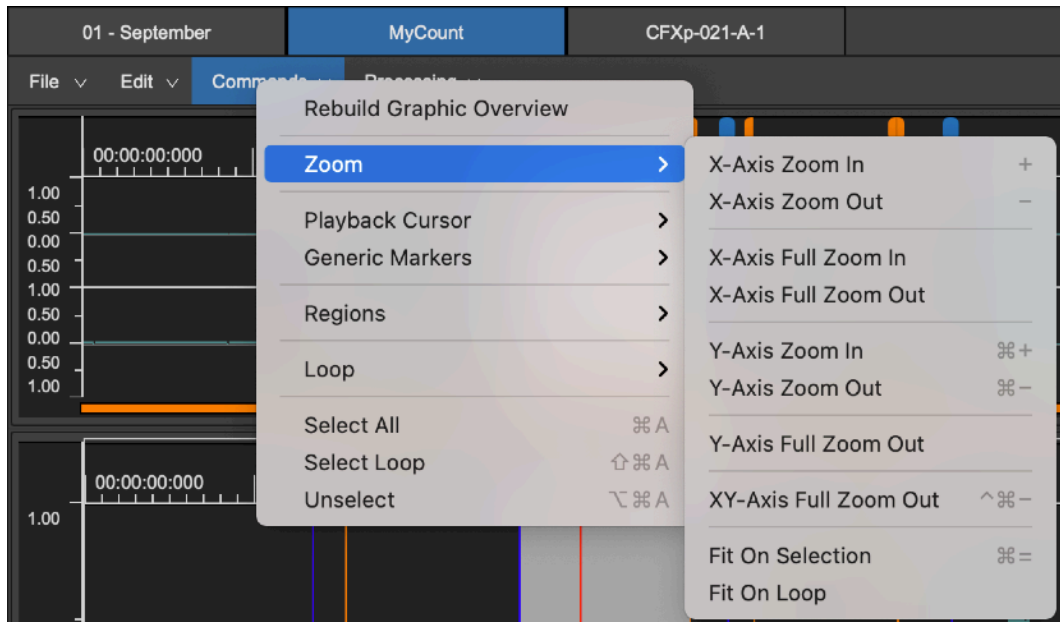
Trim Selection (or ⌘T):

to cut the regions outside the selection on the waveform view. If there is not a selection, this command is not available.

NOTE: the option on Preferences to keep a certain number of samples doing cuts is used also by the TRIM command.

The Commands Menu

Clicking on the Commands button on top of the waveform view, DSP-AudioEditor shows a menu with various commands, among which the ones for managing the markers, regions and the Loop of the audio file:



Rebuild Graphic Overview...

Doing edits as cut, copy, insert and similar operations, DSP-AudioEditor rebuilds the waveform overview using a very fast algorithm. This method can, after several cuts or pastes, cause a small approximation on the time x-axis alignment of the displayed waveform. In this case, the waveform can appear out of sync of some pixels when zoomed-out. Use this command to fix the situation rebuilding the whole waveform overview. If the file is very long, it requests some seconds to complete but, after this, the overview is again perfectly aligned.

NOTE: it is recommended to use this command every 20-30 operations of cut, copy, insert or similar.

Zoom...

Clicking on zoom opens a sub menu with zoom commands to zoom in/out on both X and Y axis.

X-Axis Zoom In (or +):

to zoom-in the waveform view on the X-Axis by a factor x2.

X-Axis Zoom Out (or -):

to zoom-out the waveform view on the X-Axis by a factor x2.

X-Axis Full Zoom In:

to full zoom-in the waveform view on the X-Axis.

X-Axis Full Zoom Out:

to full zoom-out the waveform view on the X-Axis.

Y-Axis Zoom In (or ⌘+):

to zoom-in the waveform view on the Y-Axis by a factor x2.

Y-Axis Zoom Out (or ⌘-):

to zoom-out the waveform view on the Y-Axis by a factor x2.

Y-Axis Full Zoom Out:

to full zoom-out the waveform view on the Y-Axis.

XY-Axis Full Zoom Out:

to full zoom-out the waveform view on both the Y and X Axis.

Fit on Selection (or ⌘=):

to zoom-in or zoom-out on the Y-Axis the waveform view such to make visible the current waveform selection. This command is not available if there is not a waveform selection.

Fit on Loop:

to zoom-in or zoom-out the waveform view on the Y-Axis such to make visible the Loop. This command is not available if the Loop is OFF.

NOTE: It is possible to **use the mouse wheel** to Zoom In/Out on the X-Axis. The same can be done moving vertically 2 fingers on the Apple Trackpad, if any. Use **⌘ + mouse wheel** to set the vertical Y-Axis zoom-in/out ratio.

Playback Cursor...

Clicking on Playback Cursor opens a sub menu with the commands to place the playback cursor (RED marker on the waveform view) on specific positions.

Please refer to the previous chapter about **Working with Markers** for a full reference of this menu commands.

Generic Markers...

Clicking on Generic Markers opens a sub menu with the commands to manage the Generic markers.

Please refer to the previous chapter about **Working with Markers** for a full reference of this menu commands.

Regions...

Clicking on Regions opens a sub menu with the commands to manage the audio regions of the audio file.

Please refer to the previous chapter about **Working with Markers** for a full reference of this menu commands.

Turn Loop ON/OFF... (or ⌘L)

To toggle the Loop ON and OFF. If the Loop has been turned ON, the two Loop markers (the yellow markers) will appear on the waveform view. Otherwise, DSP-AudioEditor will hide them.

See also the previous chapter about **Working with the Loop**.

Lock Loop Length... (or ⌘⇧L)

To lock/unlock the length of the Loop. If the Loop length is locked, moving any one of the two Loop markers, the other one will move accordingly. This option is useful in the case that, once you have found a good Loop, you wish to shift the Loop without changing its length. This command is not available if the Loop is OFF

See also the previous chapter about **Working with the Loop**.

to place the Loop markers at the current selection start and end. (This command is not available if there is not a selection on the waveform view). If the Loop is turned Off, it will automatically be turned On.

More about this in the previous chapter about **Working with the Loop**.

Show/Hide Loop Window (or ⌘L):

to show or hide the Loop floating window tool.

More about this in the previous chapter about **Working with the Loop**.

Select All...

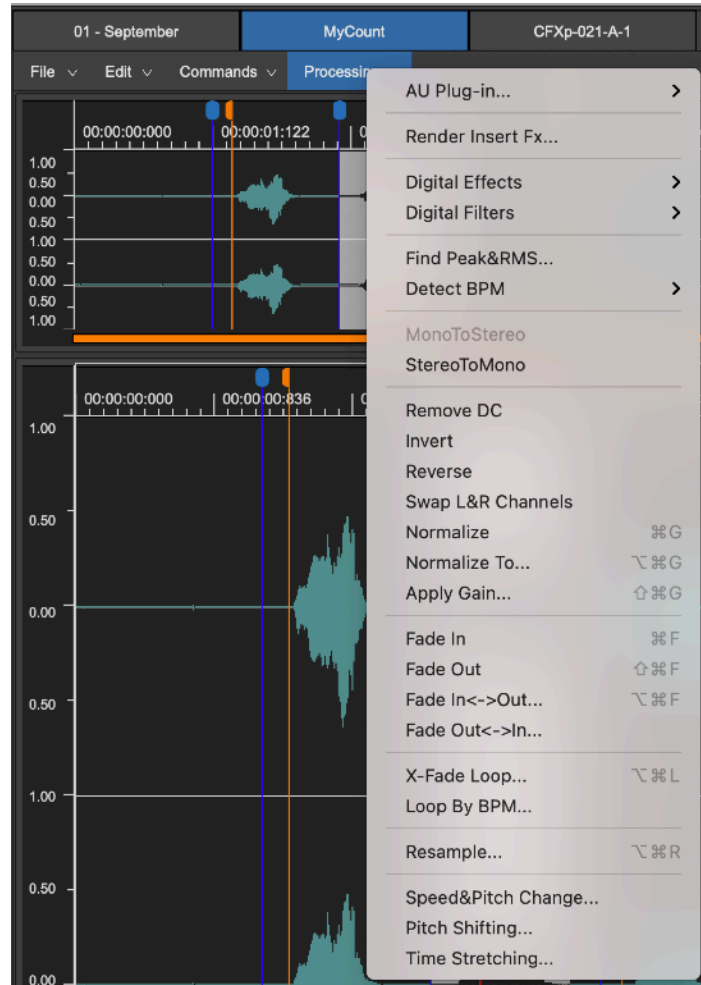
To select all the audio file on the waveform view.

Select Loop...

To select the Loop on the waveform view. This command is not available if the Loop is OFF.

The Processing Menu

Clicking on the Processing button on top of the waveform view, DSP-AudioEditor shows a menu with the commands for processing the audio file:



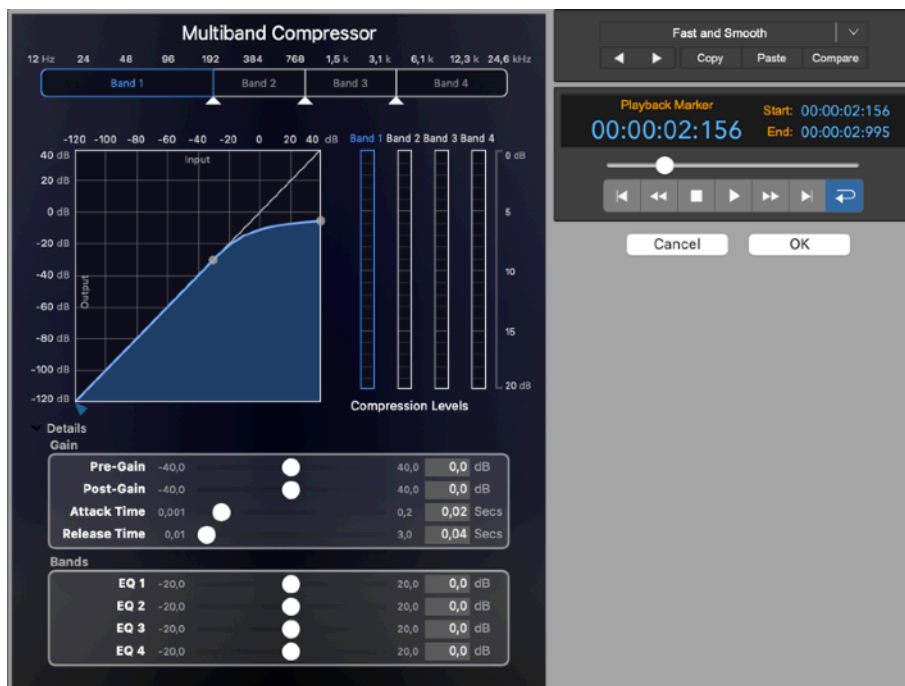
AU Plug-in...

Clicking on this menu item will open a sub menu on which DSP-AudioEditor lists all Audio Unit (AU) plug-ins installed on MacOS, grouped by manufacturers.

DSP-AudioEditor asks to MacOS the list of the installed plug-ins. To be visible, the AU plug-in must be a 64 bits effect plug-in.

Selecting one of the available plug-ins, DSP-AudioEditor opens a dialog displaying the custom interface of the AU plug-in (it depends on the plug-in itself) on the left part and, on the right, a preview controller to manage the plug-in preset and with a transport bar to preview the result before to apply it. DSP-AudioEditor will automatically resize the dialog accordingly to the plug-in custom view.

As an example, this is the dialog which DSP-AudioEditor shows loading the Apple Multiband Compressor plug-in which is built-in and available for free in MacOS:



NOTE: it is highly recommended to explore the excellent Apple AU plug-ins which are always present because built-in into MacOS. They offer an excellent quality for free.

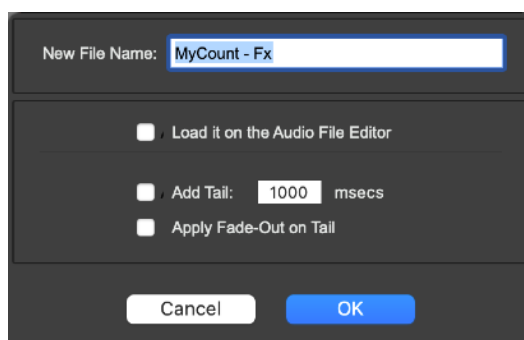
NOTE: If there is a selection on the waveform view, the function will process only the selected region, otherwise all the audio file.

Please refer to the previous chapter about the real time preview controller to know how to use it for loading/saving the current plug-in preset and for previewing the processed audio.

Render Insert FX...

Using DSP-AudioEditor is possible to load a serial chain of effect plug-ins on insert to the channel strip of the audio editor, which is different for each audio file under editing. Use this command to render this serial chain of effect plug-ins into a new audio file. The result is the same as playing the current audio file thru the effects chain and then saving the result into a new file which DSP-AudioEditor loads automatically on the Document Manager and - optionally - opens into a new tab of the waveform Editor view.

Clicking on this menu command, DSP-AudioEditor shows a dialog to set the parameters for rendering the plug-ins loaded on the channel strip into a new audio file:



New File Name:

type the new file name.

Load it on the Audio File Editor:

if turned ON, DSP-AudioEditor, after processing and loading the new processed audio file into the Document Manager, opens the new file into a new tab of the waveform Editor view.

Add Tail:

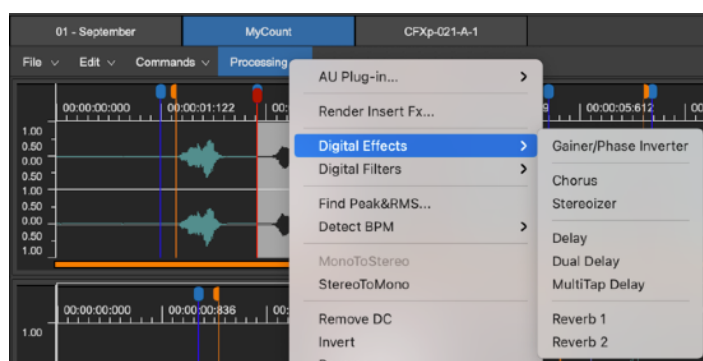
It is possible to add a tail to the destination audio file, setting also how much this tail must be long. This is very useful if the FX chain includes an effect like a reverb or space simulator which produces an additional tail after stopping the source audio file at its end. Turning ON the option “Add Tail”, DSP-AudioEditor will continue to process the audio after reaching the audio file end, capturing the tail of the effect or of the chain of effects.

Apply Fade-Out on Tail:

If the option to Add Tail is ON, it is possible to apply a fade-out curve to the tail, avoiding an abrupt end at the end of the destination file. Turning ON “Apply Fade-Out on Tail”, DSP-AudioEditor decreases the output gain while processing the tail, resulting in a smooth end.

Digital Effects...

DSP-AudioEditor has several high quality digital effects which are built-in into the application itself. They can be seen as custom plug-in which are unique to DSP-AudioEditor.



NOTE: If there is a selection on the waveform view, the function will process only the selected region, otherwise all the audio file.

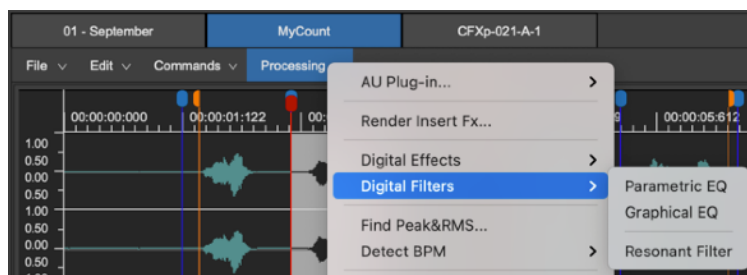
This group of effects includes the effects using modulated delay lines and chamber/hall simulations: Chorus, Stereoizer, Delay, Dual Delay, Multitap Delay, Reverbs 1&2.

More about using these effects in the chapter about Using the built-in Effects of DSP-AudioEditor.

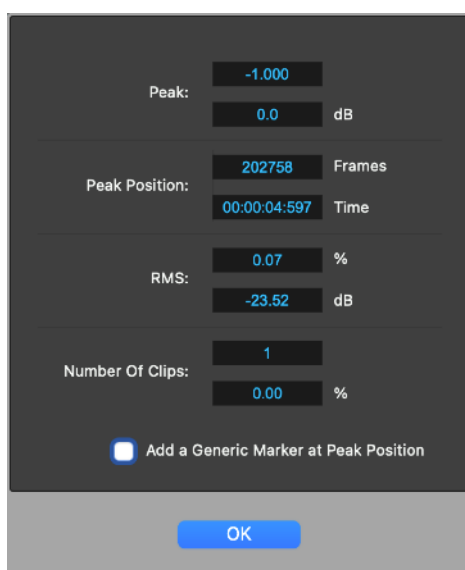
Digital Filters...

This group of effects includes the digital filters: Parametric & Graphic Equaliser, Resonant filters (Analog Filter Emulators).

More about using these effects in the chapter about Using the built-in Effects of DSP-AudioEditor which is later in this manual.



Find Peak&RMS...



This command opens a dialog on which DSP-AudioEditor shows the Peak and RMS level computed on the audio file. If there is a waveform selection on the waveform view, DSP-AudioEditor shows the Peak and RMS level computed only on the selected region.

DSP-AudioEditor shows also the number of clips, defined as the points where the waveform reaches the maximum values of -1.0 or 1.0 , and the ratio between the number of clips and the number of frames of the audio file.

It is also possible to add a generic marker at the position of the detected peak (or at the position of the first clip, if there are more audio clips). If you want to add a marker, turn ON the check box.

Detect BPM...

This menu item opens a sub menu on which you can choose the region used by DSP-AudioEditor to calculate the Beats Per Minute (BPM).

These are the different options:

Use All

DSP-AudioEditor calculates the Beats Per Minute (BPM) using all the audio file length.

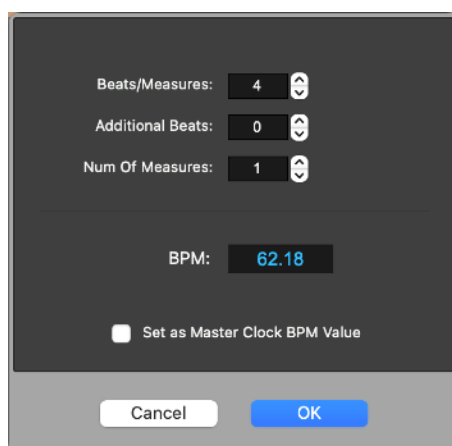
Use Selection

DSP-AudioEditor calculates the Beats Per Minute (BPM) using the length of the selection on the waveform view. This command is not available if there is not a selection.

Use Loop

DSP-AudioEditor calculates the Beats Per Minute (BPM) using the length of the Loop. This command is not available if Loop is OFF.

DSP-AudioEditor shows the following dialog:



To calculate the BPM value, it is necessary to enter:

- **Beats/Measures:** number of Beats for each Measure. This is the number of metronome hits. As an example, in case of a time signature of 3/4, it is equal to 3.

- **Additional Beats:** it is the rest of Beats.

- **Num Of Measures:** it is the total number of measures.

For example, in case we want to know the BPM from a region equal to 3 measures in 4/4, plus one in 3/4, we need to set:

Beats/Measures	= 4
Additional Beats	= 3
Num of Measures	= 3

- **BPM** shows the calculated result. When one of the values in the upper part changes, the value of the BPMs updates in real time accordingly.

- **Set As Master Clock BPM Value:** check this flag to set the calculated BPM value as Master BPM Tempo value. It is the master clock use by DSP-AudioEditor as reference for sending the sync message to AU plug-ins, effects.

MonoToStereo

Given a mono audio file, this command change it to a stereo file duplicating the single audio channel.

This command is not available if the audio file is stereo.

StereoToMono

Given a stereo audio file, this command change it to a mono file mixing the left and right channels into a single audio channel. The source left and right channels will be gained by a factor $\times 0.5$ before mixing to avoid clipping.

This command is not available if the audio file is mono.

Remove DC...

To subtract (remove) the DC Level from an audio file.

The DC Level of an audio file is the average value of all samples. The waveform that represents the audio file should oscillate around the zero, and therefore its average value should be zero. But it can happen that the waveform oscillate around a value (DC) which is not the zero level. This can be caused by, for example, a bias current in the analog circuit of the Analog To Digital (A/D) converter used for sampling the signal.

If there is a selection, this function process only the selected region, otherwise all the file.

Invert...

To invert the audio signal around the zero level, ie to change the sign of all the samples in the region to edit.

If there is a selection, this function process only the selected region, otherwise all the file.

Reverse...

To reverse in time the audio signal of the audio file, playing it on reverse, from the end to the beginning.

If there is a selection, this function process only the selected region, otherwise all the file.

Swap L&R Channels...

To swap the samples between left and right channels of a stereo audio file.

If there is a selection, this function process only the selected region, otherwise all the file.

Normalize... (or ⌘G)

To apply an automatic linear gain to samples of the audio file such that the level of maximum peak of the processed audio file will be equal to 0 dB.

NOTE: this is one the most used editing function. Doing a recording, usually it is safer to keep the recording level low enough to avoid audio clips at 0 dB, which would cause a distortion. Then it is a good practice to apply this edit function to bring the level of the signal to the maximum allowed level.

If there is a selection, this function process only the selected region, otherwise all the file.

Normalize To... (or ⌘G)

as the above Normalize processing function, with the difference that the target maximum peak of the processed audio file is a user programmable value.

Then, DSP-AudioEditor shows a dialog where:



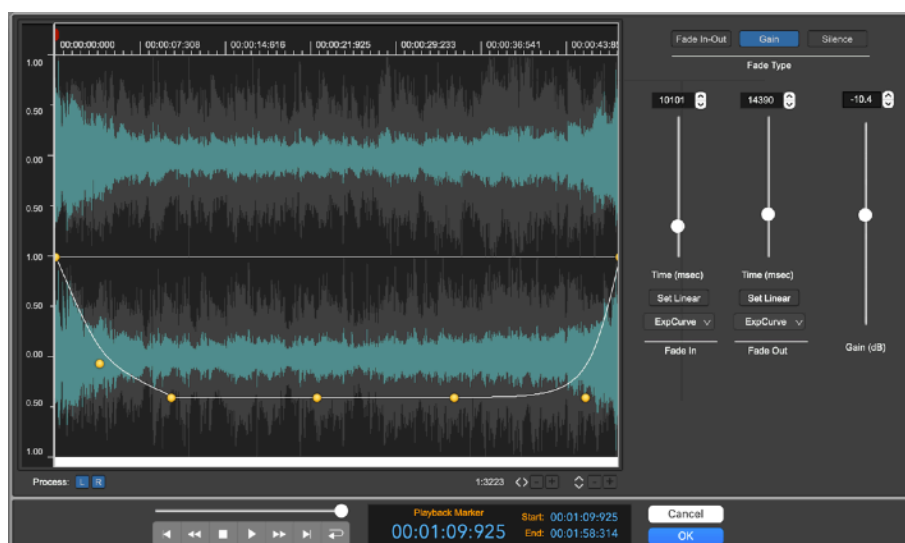
On the right side, there are the results of the analysis of the region of the audio file to process. These results are the same as seen running the Find Peak&RMS command.

On the left a slider to set the target for the peak of the audio file after processing.

If there is a selection, this function process only the selected region, otherwise all the file.

Apply Gain...

This command opens the **Fade Editor Tool** setting the Fade Type to Gain by default.



The Fade Editor is a graphical tool used to apply a gain on a certain region defining also user programmable curves for the fade-in and fade-out for smoothing the gain changes.

Please refer to the chapter describing the Fade Editor Tool to know more about how to use it.

If there is a selection, this function process only the selected region, otherwise all the file.

Fade In... (or ⇧⌘F)

To apply a linear fade-in gain curve to the region to process. The level of the processed audio file will start from zero at region start to reach the full level at region end.

If there is a selection, this function process only the selected region, otherwise all the file.

NOTE: this is a fast shortcut to fade-in a certain region, without any graphical tool. For more complex edits, it is possible to use the Fade Editor Tool. Please refer to the chapter describing the Fade Editor Tool to know more about how to use it.

Fade Out... (or ⇧⌘F)

To apply a linear fade-out gain curve to the region to process. The level of the processed audio file will start from the full level at region start to reach the zero at the region end.

If there is a selection, this function process only the selected region, otherwise all the file.

NOTE: this is a fast shortcut to fade-out a certain region, without any graphical tool. For more complex edits, it is possible to use the Fade Editor Tool. Please refer to the chapter describing the Fade Editor Tool to know more about how to use it.

Fade In<->Out... (or ⌘⌘F)

This command opens the **Fade Editor Tool** setting the Fade Type to Fade In-Out by default.

If there is a selection, this function process only the selected region, otherwise all the file.

Please refer to the chapter describing the Fade Editor Tool to know more about how to use it.

Fade Out<->In...

This command opens the **Fade Editor Tool** setting the Fade Type to Silence by default.

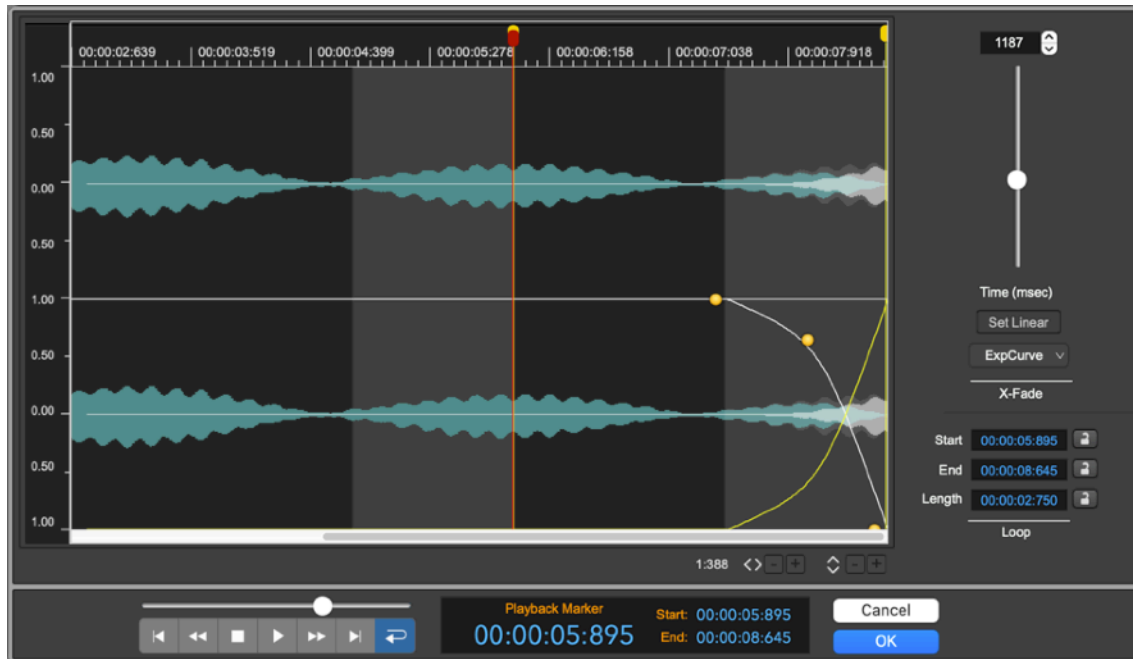
If there is a selection, this function process only the selected region, otherwise all the file.

Please refer to the chapter describing the Fade Editor Tool to know more about how to use it.

X-Fade Looping... (or ^⌘L)

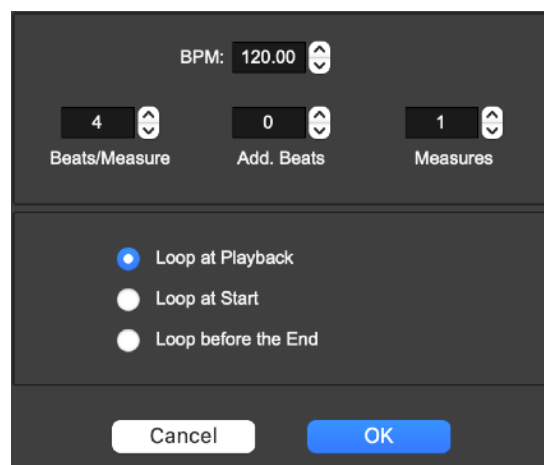
- This command opens the **X-Fade Looping Editor**.

The X-Fade Looping Editor is a graphical tool used to apply a X-Fade looping algorithm to the Loop region. It is possible to define the shape of the X-Fade curves, checking visually what happens to the processed audio file, previewing the waveform and listening to the result as well.



Please refer to the chapter describing the X-Fade Looping Editor to know more about how to use it.

Loop by BPM...



To set the Start and End Loop points such that its length is calculated from the value expressed as Beats Per Measures (BPM). It is also possible to set if the Loop must start at the beginning of the audio file, at the playback cursor, or if it must end at the end of the audio file.

NOTE: This is a very useful function when you want to set a Loop on rhythmic patterns. In this case the length of the Loop has to be based considering the Beats Per Measures metronomic tempo.

It is necessary to set the following parameters:

- **BPM:** it is the value of the target Beats Per Measures (BPM)
- **Beats/Measures:** number of Beats for each Measure. This is the number of metronome hits. As an example, in case of a time signature of 3/4, it is equal to 3.
- **Additional Beats:** it is the rest of Beats.
- **Measures:** it is the total number of measures.

As an example, in case we want to position the Loop points on a region equal to 3 measures in 4/4, plus one more in 3/4, with a BPM time of 120.32, we must set:

BPM	= 120.32
Beats/Measures	= 4
Measures	= 3
Add. Beats	= 3

It is also necessary to set where the Loop must start:

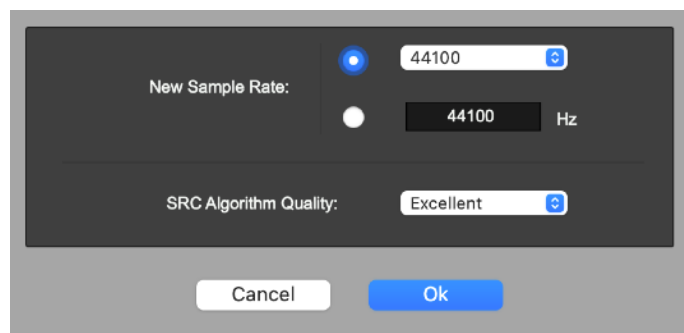
- **Loop At Playback :** the Loop starts at the playback cursor
- **Loop At Start :** the Loop starts at the beginning of the audio file
- **Before End :** the Loop ends at the end of the audio file

If the Loop was OFF, DSP-AudioEditor also turns it ON automatically.

Resample... (or ⌘R)

To change the sample rate of the audio file by using a HQ linear phase Sample Rate Converter (SRC) algorithm of DSP-AudioEditor.

The Nyquist theorem demonstrates that the maximum frequency which can be reconstructed from a digital audio signal is the half of its sample rate. For this, lowering the sample rate, the SRC algorithm must also remove the highest frequencies components to guarantee that the maximum frequency of the digital audio signal is lower than the half of its sample rate. This will lower the audio spectrum range of the audio signal. The benefit is that the audio file will have smaller dimensions, saving space on the hard disk.



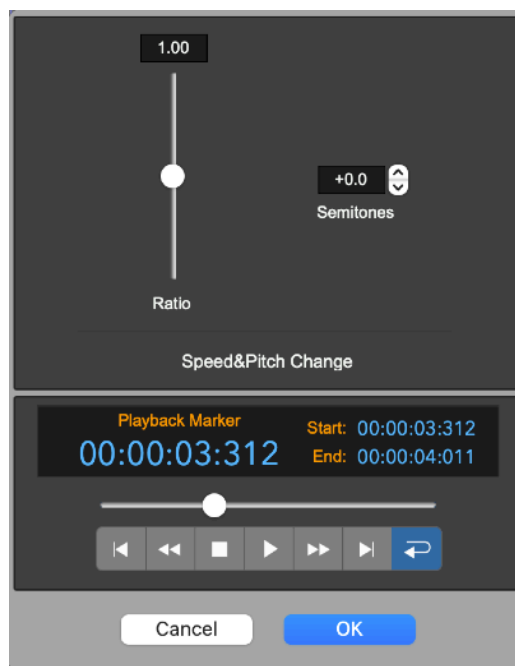
NOTE: the offline Sample Rate Converter (SRC) algorithm of DSP-AudioEditor uses a very High Quality **Linear Phase Sample Rate Converter (SRC)** Resampling algorithm, based on the state-of-art technology about the Digital Signal Processing and linear phase digital filtering. This SRC guarantees the maximum audio quality, in terms of frequency response and audio quality (the filter characteristic is outstanding and being Linear Phase, the algorithm does not introduce phase inversion at the transition band).

- **New Sample Rate:** to set the sample rate of the processed audio file. It is possible to choose one of the most common values on the popup menu or to enter the new value numerically.
- **SRC Algorithm Quality:** to set the quality of the offline Sample Rate Converter (SRC) algorithm. The Excellent quality needs more CPU time than the others. It will produce the result with the maximum quality but will be slower to complete the process. Possible values are Fast, Good, Better, Excellent.

Speed&Pitch Change...

To change the pitch of the audio file playing it faster (to rise the pitch) or slower (to lower the pitch). This is equivalent to modify the speed of a tape recorder. Also the length of the audio file will change accordingly. Playing it faster, it will take less time to reach the end, producing a shorter audio file, the opposite playing it slower.

DSP-AudioEditor shows the following dialog:



- **Ratio:** to set the ratio between the pitch of the processed audio file and the source as a float value (ratio = 1.0 means no change, ratio = 2.0 means an octave up, and so on).
- **Semitones:** to set the ratio between the pitch of the processed audio file and the source as semitones (semitones = 0.0 means no change, semitones = +12.0 means an octave up, and so on).

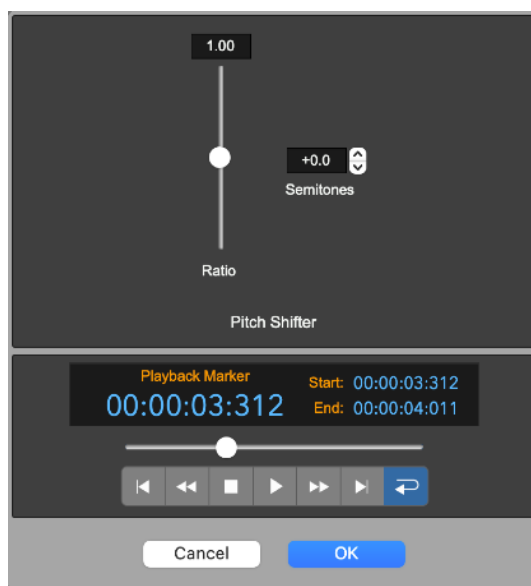
If there is a selection, this function process only the selected region, otherwise all the file.

NOTE: It is also possible to modify the pitch without change the speed - or time (duration of the audio file), using the Pitch Shifting command of DSP-AudioEditor.

Pitch Shifting...

To change the pitch of the audio file without changing the speed (duration of the audio file). This algorithm does not modify the time - length - of the audio file.

DSP-AudioEditor shows the following dialog:



As for the Speed&Pitch Change command,

- **Ratio:** to set the ratio between the pitch of the processed audio file and the source as a float value (ratio = 1.0 means no change, ratio = 2.0 means an octave up, and so on).
- **Semitones:** to set the ratio between the pitch of the processed audio file and the source as semitones (semitones = 0.0 means no change, semitones = +12.0 means an octave up, and so on).

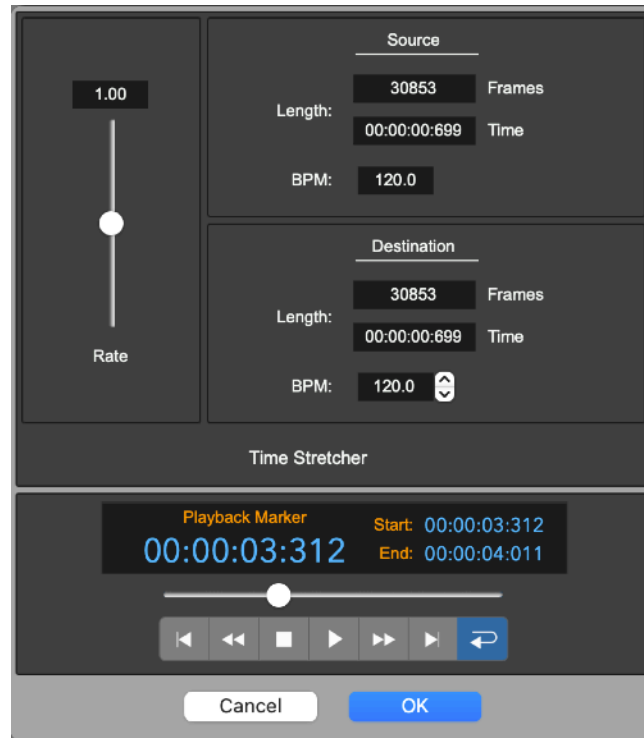
If there is a selection, this function process only the selected region, otherwise all the file.

NOTE: To achieve the result, DSP-AudioEditor uses a very High Quality Spectral Re-Synthesizer, which is a very sophisticated algorithm compared to the algorithm used for Speed&Pitch Change.

Time Stretching...

To change the speed (time) of the audio file without changing the pitch.

DSP-AudioEditor shows the following dialog:



There are different options to set the ratio between the length of the destination and of the source file:

- **Ratio:** it is possible to set the ratio using the slider on the left, or using the numerical control above the slider as well.
- **Destination Length:** it is possible to enter directly the length of the destination audio file, expressing it as total number of frames or as total time.
- **Destination BPM:** it is also possible to set the ratio expressing it as Beats Per Minutes (BPM), which is a more usual form used by musician. For this, first it is necessary to set the BPM of the source file, clicking on the SET button in the section related to the parameters of the source file. This opens a dialog that allows you to set all the parameters needed to compute the source BPM, as previously described for the Detect BPM command. Then, use the numerical control to set the destination BPM in the section related to the parameters of destination file. DSP-AudioEditor will automatically set the time stretching ratio.

If there is a selection, this function process only the selected region, otherwise all the file.

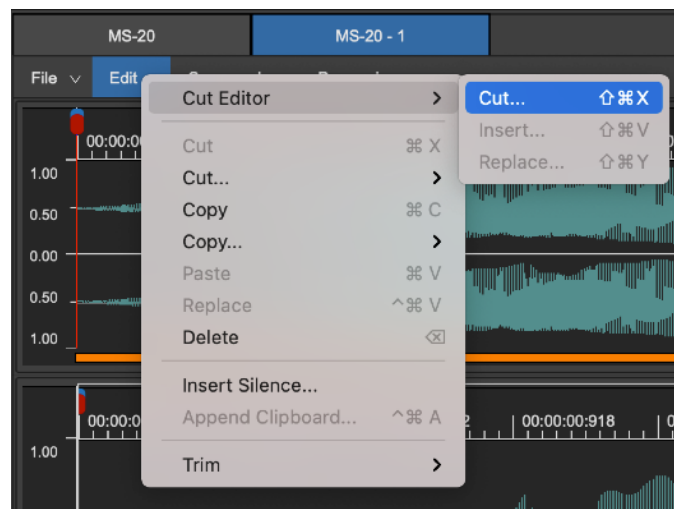
NOTE: DSP-AudioEditor uses an High Quality Spectral Re-Synthesizer doing Time Stretching.

The Cut Editor Tool

DSP-AudioEditor offers a very unique Cut Graphic Editor tool to make cuts on the audio file, as well as pastes of the clipboard into an audio file, without any problem caused by possible abrupt changes in the level and slope of waveforms at cut or paste points.

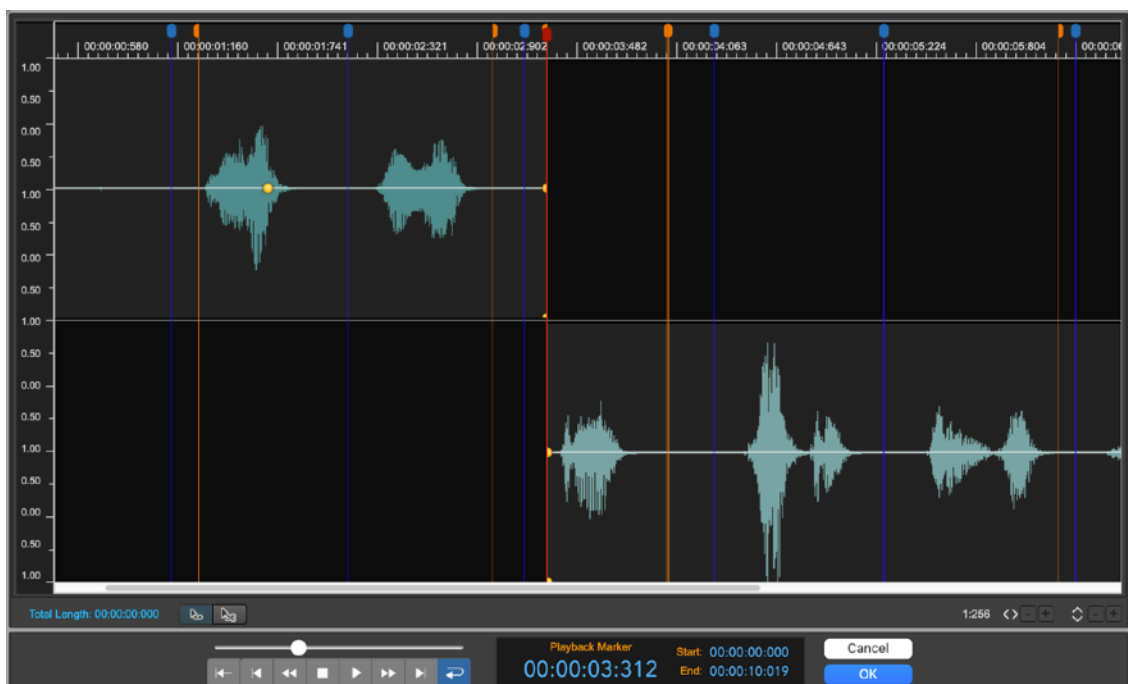
Doing a graphical Cut using the Cut Editor Tool

To start using the Cut Editor, click on the Edit button on top of the waveform view, and select Cut Editor, DSP-AudioEditor shows a submenu with the commands for opening the Cut Editor:



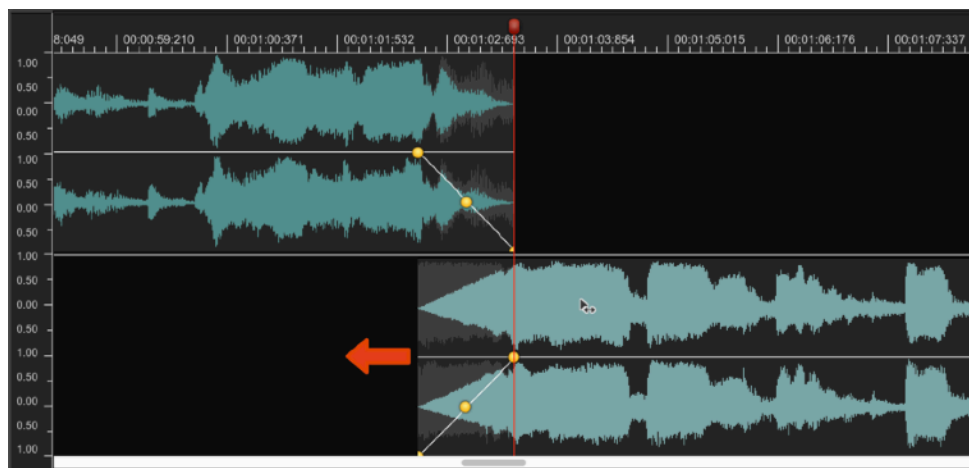
Cut... (or ⌘X):

DSP-AudioEditor opens the Cut Editor to do a cut at playback position:



In the upper part, to the left, there is the waveform before the cut point. In the lower part, to the right, there is the waveform after the cut point.

Click on the waveform on the right, that's the audio file region after the cut point, and drag it to the left such to overlap the waveform in the upper part, that's the region before the cut point. DSP-AudioEditor will automatically create a cross-fade curve between the two overlapping regions:

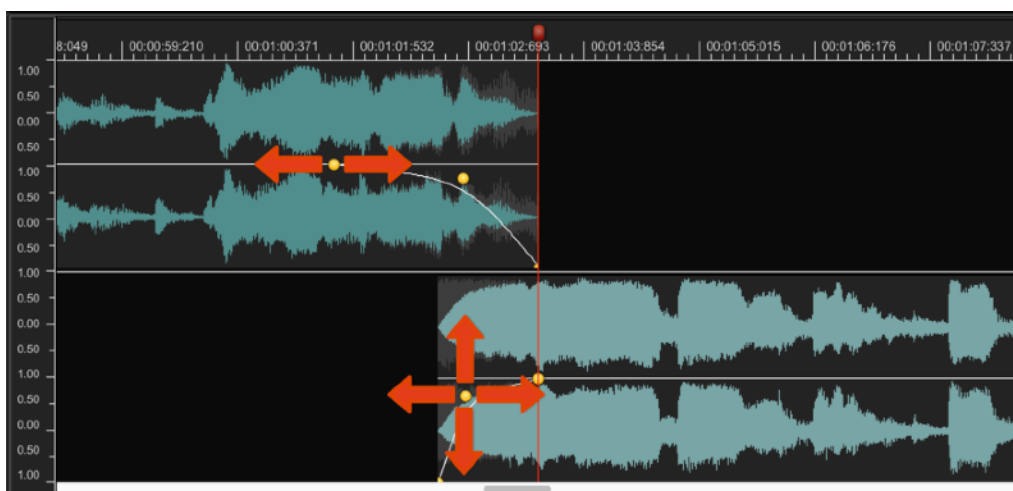


Setting fade-In/Out lengths and shapes of the curve

The Cut Editor Tool gives the complete freedom for setting the cross-fade lengths and the shape of the fade-in/out curves. Fade-In length and curve is independent from Fade-Out length and curve, if necessary.

To set the length only of the fade-out of the upper waveform, click with the mouse on the little yellow circle control at the beginning of the fade-out, and shift it backward or forward along the time axis. Do the same procedure to set the fade-in length of the lower waveform, that's the audio file region after the cut point.

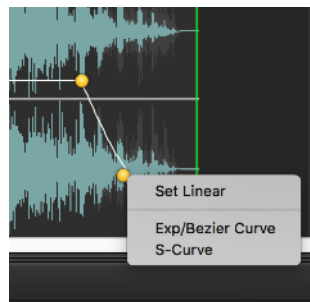
To set the shape of the curve of the fade-out, click on the little circle control in the middle of the fade-out curve, and drag it to the four directions setting the slope which best suite your needs. Do the same to set the shape of the curve of the fade-in of the waveform after the cut point.



DSP-AudioEditor allows two different types of shapes for the Fade-In/Out curve:

- **Exponential/Bezier** type: in this case the shape control point can move on all directions, setting all the shapes ranging from logarithmic, linear, exponential.
- **S-Curve** type: in this case the shape control point can move only up or down, the shape is convex or concave according to the vertical position of the shape control point.

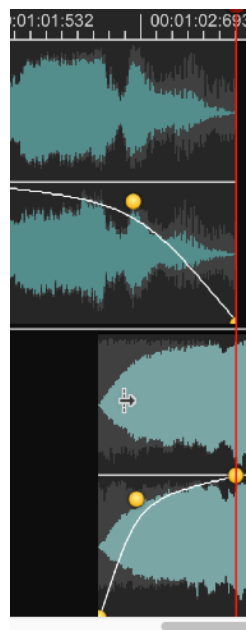
To set the type of the shape, click with the right mouse button on the shape control point, DSP-AudioEditor will show an utility popup menu with the command:



NOTE: by default, the shape of each curve is independent. Keeping the **SHIFT key** pressed while clicking and dragging the control to set the shape of a the fade-In or fade-Out curve, then also the other fade curve will follow such that the sum of the gain of the two curves is always constant (equal power).

Trimming the the start and end of the region at cut point

Moving the mouse to the border of the upper waveform or of the bottom waveform, the mouse mode will change to the Trimmer:



In this mode, clicking and dragging the mouse to the left or to the right, DSP-AudioEditor trims the region end (trimming the waveform before the cut point) or region start (trimming the waveform after the cut point).

Inserting the Clipboard or Replacing with the Clipboard using the Cut Editor Tool

On the Cut Editor sub menu, if the Audio clipboard of DSP-AudioEditor is not empty, these other commands are also available:

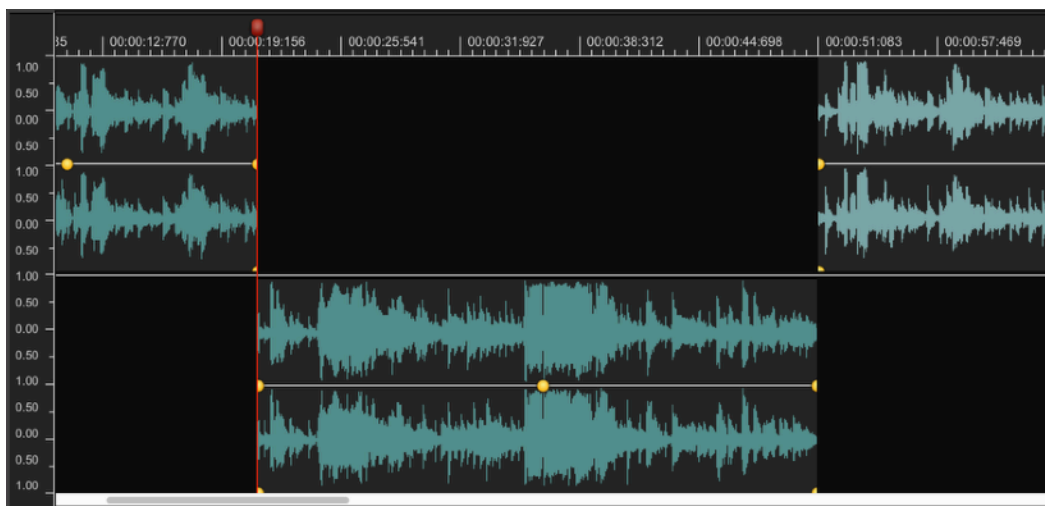
Insert... (or ⌘⌘V):

DSP-AudioEditor opens the Cut Editor, do the cut at playback position, and inserts the clipboard at the position of the cut. In the upper part, to the left, there is the waveform before the cut point. In the lower part, in the middle, there is the waveform of the audio clipboard then, again in the upper part, there is the waveform starting just after the cut point.

Replace... (or ⌘⌘Y):

DSP-AudioEditor opens the Cut Editor, do the cut at playback position, and places the clipboard at the position of the cut. In the upper part, to the left, there is the waveform before the cut point. In the lower part, in the middle, there is the waveform of the audio clipboard then, again in the upper part, there is the source audio file waveform without the presence of the clipboard in between.

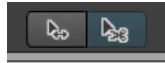
The difference between Inserting and Replacing is that in the first case DSP-AudioEditor makes the space to place the clipboard at cut position, shifting the rest of the source audio file forward along the time axis. In the second case it keeps in sync the last section of the source audio file, without shifting it along the time axis.



Also in this case, it is possible to shift the waveforms of the clipboard and of the region after the cut along the time axis. It is possible to set the fade-in/out length as set the fade-in/out curve shapes as well.

Using the scissor mode to make additional cuts

The Cut Editor allows to make additional cuts and cross-fades, if needed. Clicking on the scissor button positioned to the bottom of the waveform view:



The mouse pointer will change to a scissor. Then, clicking on the waveform view, the Cut Editor splits the audio file exactly on that position, moving the waveform which follow the cut point on the other side of the view. It is possible to repeat the operation as many times as necessary.

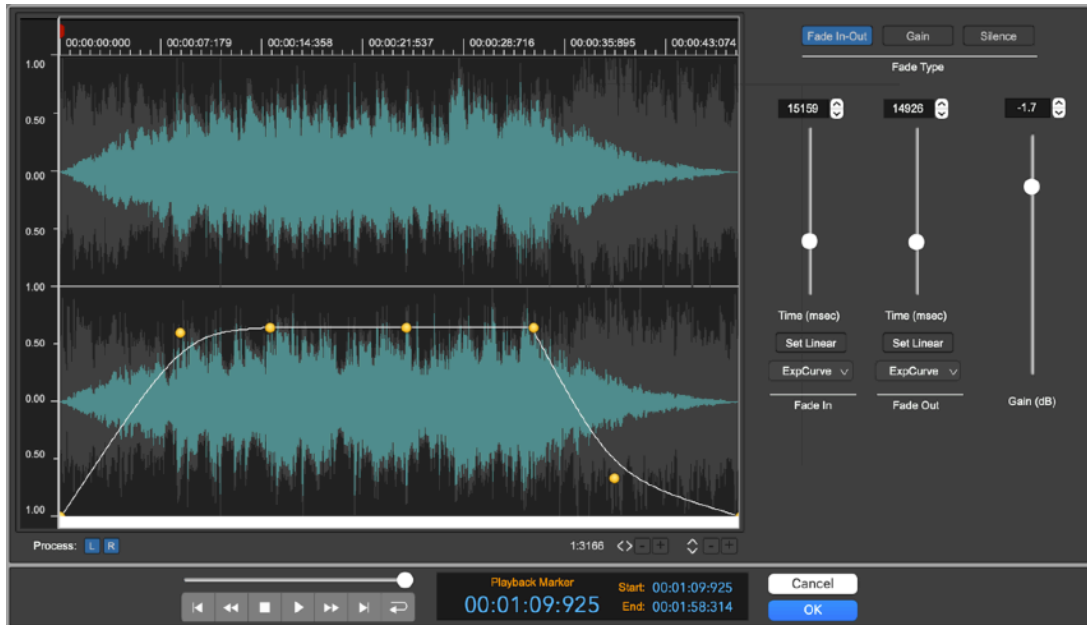


To return to be able to shift the waveforms on the waveform view, click on the shifter button which is positioned to the right of the scissor button.

Zooming In/Out is possible as well. Use the mouse wheel, the keyboard or the buttons on the lower right part of the Cut Editor for zooming.

The Fade Editor Tool

The commands **ApplyGain...**, **FadeIn<->Out...** and **FadeOut<->In...**, available on the Processing menu, use the same Fade Editor tool window:



On the left part, the Fade Editor shows the graphical representation of the envelope curve composed by a Fade-In, a Gain level and a Fade-Out. It is possible to set the length as well as the shapes of the FadeIn/Out curves clicking and dragging the fade yellow circular controls.

On the right part, the Fade Editor shows sliders and editable fields to enter the fades and gain values numerically.

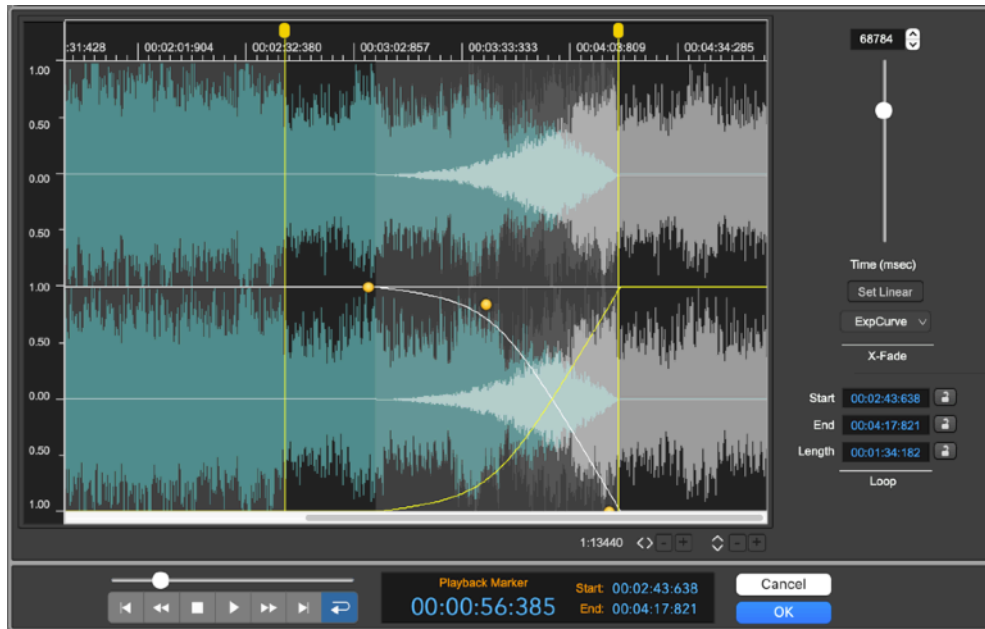
On the upper right part of the the Fade Editor, the possible settings of the Fade Types are:

- **Fade In-Out** (automatically set by calling the **FadeIn<->Out...** command): in this case the envelope curve starts at zero gain (-inf dB), to fade-in reaching the Gain level, then fades-out to reach the zero level at the end.
- **Gain** (automatically set by calling the **ChangeGain...** command): in this case the envelope curve starts at +0dB level (no level change on the source file), fades to the Gain value, then fades to reach again the +0dB at the end. Use this option to change the gain of the source audio file without changing the audio levels at the borders.
- **Silence** (automatically set by calling the **FadeOut<->In...** command): starts at +0dB level (no change of level on the source file), fades to the 0 level (silence), then fades again to +0dB. It is similar to Gain where the Gain level is at zero (-inf dB).

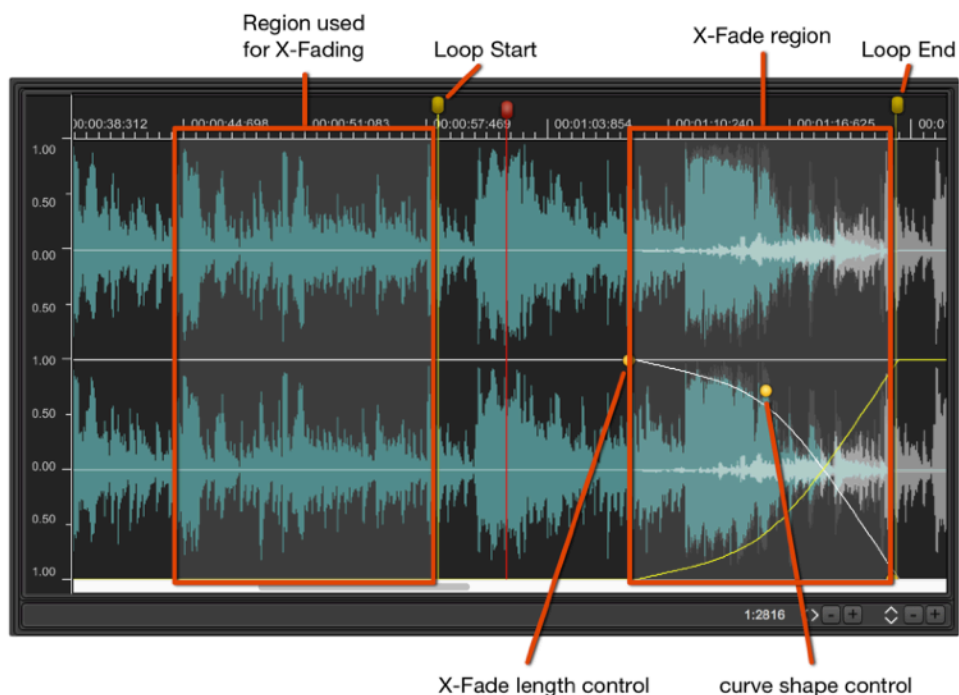
NOTE: by default, if the file is stereo, the envelope curve processes both the L&R channels. It is possible to process only one channel turning OFF the channel to keep as it is, clicking on the L or R square buttons below the waveform view, in the right part to the Fade Editor window.

The X-Fade Looping Editor Tool

The X-Fade Looping Editor is a graphical tool used to apply a X-Fade looping algorithm to the Loop region. It is possible to define the shape of the X-Fade curves checking in real time what happens to the processed audio file visually, previewing the waveform and listening to the result as well.



The X-Fade Looping algorithm, available only if the Loop is turned On, modifies the samples into the loop region mixing the region before the Loop Start with the region before the Loop End, applying a cross-fade (X-Fade) between the samples to copy and the samples already on that region.



The purpose to make the loop which plays as much as natural and as smooth as possible. Because, applying the X-Fade looping, the sample at the Loop End will be mathematically identical to the one at the Loop Start, there is not any abrupt change or audio click when playing and jumping from the loop end to the loop start. But it can happen that the result is not musical.

The success of the operation depends on several factors:

- before to apply the X-Fade, the loop points must be already positioned in places considered good for looping, even if the audio clicks at loop.
- The length of the X-Fade is a very important factor to achieve an acceptable result, and it is difficult to give in advance a precise rule about the best value to set. Generally, when applying the X-Fading Loop algorithm to short loops (less than 1 sec), it is recommended to enter high length values for the X-Fade, even equal to the full loop length. Instead, in case of long loops, smaller values for the X-Fade length are generally very much more appropriate.

The maximum value for the X-Fade length cannot never exceed the length of the loop, and it cannot exceed also the position of the loop start marker respect the beginning of the file. In other words, the maximum length of the X-Fade is smallest value between the length of the loop and the position of the loop start marker.

The Built-in Effects of DSP-AudioEditor

DSP-AudioEditor has a series of built-in digital effects of very high quality. they range from the various uses of modulated digital delay lines, to the simulation of analog filters and bank of filters. Algorithms for the room and hall simulations are available as well.

It is possible to use the built-in effects of DSP-AudioEditor in the same way as all the other Audio Unit (AU) Effect plug-ins installed on MacOS.

Using an effect as edit function in the Audio Editor, the function modifies only on the selected region if there is a selection, otherwise all the audio file.

Among these different effects, there are several shared graphic objects, such as knobs, sliders, and Vu-Meters.

This chapter describes in detail all their characteristics.

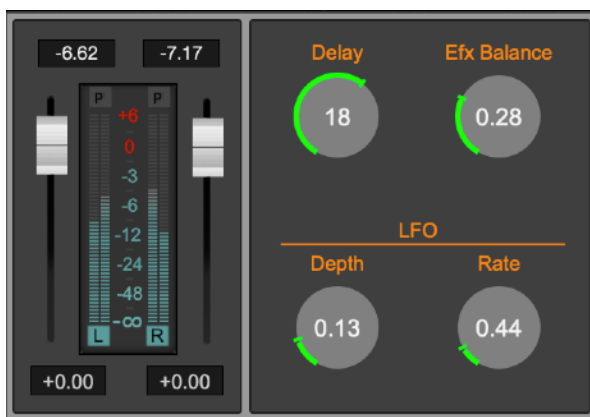
The DSP-AudioEditor built-in effects can be grouped into two main categories:

- **Digital Effects:** these are the effects using modulated delay lines and room/hall simulations (Delay, Chorus, Reverb)
- **Digital Filters:** these are the effects using digital filters (Parametric & Graphic Equalizer, Analog Filter Emulators).
- **Tools:** there are special plug-ins. One is a Gainer/Phase Inverter, one is a very special Send&Returns to/from the hardware audio device. More about this later in this chapter.

As in the case of the other AU Effect plug-ins, these effects can be used as real time effects, loading them on insert on the channel strip, or can be used for processing as any other edit function, calling them on Processing menus.

Chorus

The chorus algorithm uses dual delay lines, modulated by a Low Frequency Oscillator (LFO).



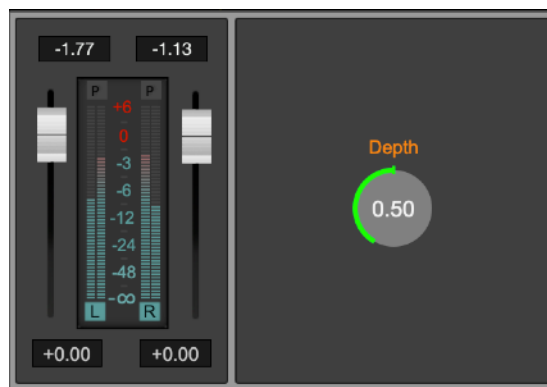
- **L&R Gain Sliders:** on the left there is a section with two gain sliders, one for the Left, one for Right channel. Use these sliders to control the gain of the input signal. Peak and RMS Vu-Meters show the input L&R signal levels.

- **Efx Balance [0.0...1.0]:** to set the ratio between the processed and the input audio signals.
- **Delay [0.0...30.0 msec]:** to set the delay time used by the algorithm. Higher values produce a more noticeable and deep chorusing.
- **LFO Depth [0.0...1.0]:** to set the amount of the modulation of the delay time by the LFO (Low Frequency Oscillator). Higher values produce more noticeable and deep chorusing.
- **LFO Rate [0.0...5.0 Hz]:** to set the speed of the LFO. Lower values produce a lighter and less intrusive effect.

Stereoizer

The Stereoizer effect uses pass-band filters to enhance the stereo image of the input signal.

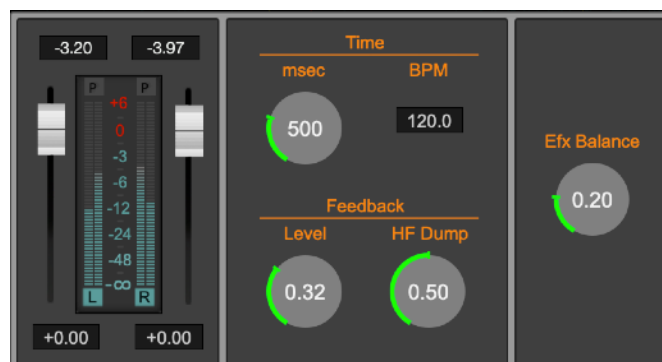
NOTE: it is highly recommended to check the mono compatibility of the signal on output, due to phase cancellation side-effect when summing the Left and Right channel if listening in mono.



- **L&R Gain Sliders:** on the left there is a section with two gain sliders, one for the Left, one for Right channel. Use these sliders to control the gain of the input signal. Peak and RMS Vu-Meters show the input L&R signal levels.
- **Depth [0.0..1.0]:** to set the depth of the effect.

Delay

Using one or more Delay effects it is possible to create very special mono and stereo echoes.



The Delay effect of DSP-AudioEditor is a single delay line, which mixes the L&R input channels before to process the signal into a delay line. It is very good to simulate the vintage sound of delay pedals used mostly by guitarists.

- **L&R Gain Sliders:** on the left there is a section with two gain sliders, one for the Left, one for Right channel. Use these sliders to control the gain of the input signal. Peak and RMS Vu-Meters show the input L&R signal levels.
- **Efx Balance [0.0...1.0]:** to set the ratio between the processed and the input audio signals.
- **Time [0..2000 msec]:** to set the time of the delay line, in msec. and in BPM (Beats Per Minute). Changing one of the values, DSP-AudioEditor will automatically change the other one in real time as well.
- **Feedback Level [0.0..1.0]:** set the feedback level of the output signal back to the input. The feedback level changes the number of repetitions (echoes) of the effect.
- **Feedback Filter Cutoff [20..16000 Hz]:** the feedback line is processed by a lowpass filter to emulate the absorption of reflecting materials, which higher for the high frequencies.

Dual Delay

The Dual Delay effect of DSP-AudioEditor uses two independent delay lines, one for the left and one for the right channel. It is very good to produce ping-pong stereo effects and to spread the stereo image of a stereo input signal.



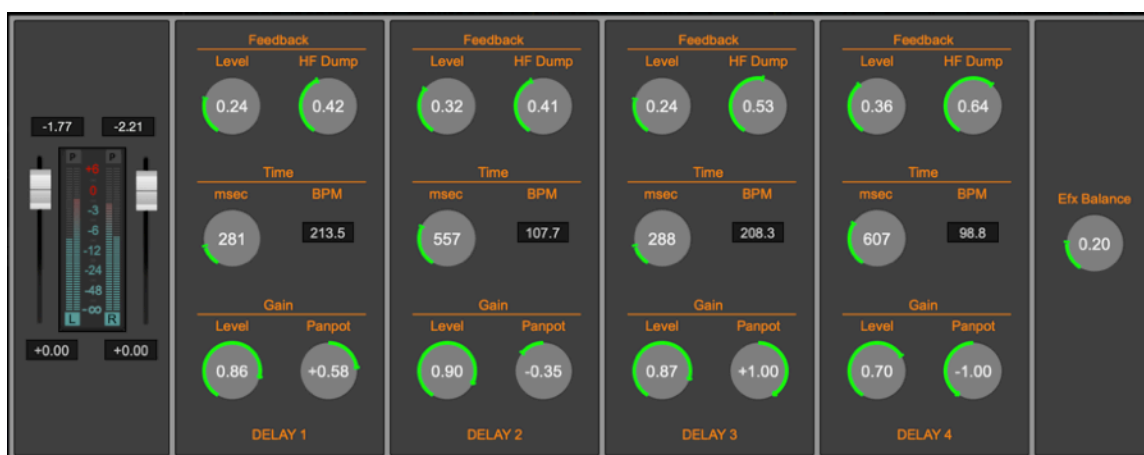
- **L&R Gain Sliders:** on the left there is a section with two gain sliders, one for the Left, one for Right channel. Use these sliders to control the gain of the input signal. Peak and RMS Vu-Meters show the input L&R signal levels.
- **Efx Balance [0.0...1.0]:** to set the ratio between the processed and the input audio signals.

For each delay line:

- **Time [0..2000 msec]:** to set the time of the delay line, in msec. and in BPM (Beats Per Minute). Changing one of the values, DSP-AudioEditor will automatically change the other one in real time as well.
- **Feedback Level [0.0..1.0]:** set the feedback level of the output signal back to the input. The feedback level changes the number of repetitions (echoes) of the effect.
- **Feedback Filter Cutoff [20..16000 Hz]:** the feedback line is processed by a lowpass filter to emulate the absorption of reflecting materials, which higher for the high frequencies.

MultiTap Delay

The MultiTap Delay effect is uses four independent delay line, each with its own controls for setting its level and pan-pot on output signal.



- **L&R Gain Sliders:** on the left there is a section with two gain sliders, one for the Left, one for Right channel. Use these sliders to control the gain of the input signal. Peak and RMS Vu-Meters show the input L&R signal levels.
- **Efx Balance [0.0...1.0]:** to set the ratio between the processed and the input audio signals.

For each delay line:

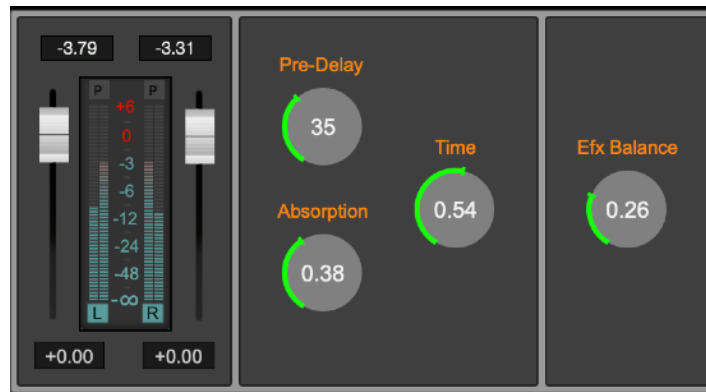
- **Time [0..2000 msec]:** to set the time of the delay line, in msec. and in BPM (Beats Per Minute). Changing one of the values, DSP-AudioEditor will automatically change the other in real time as well.
- **Feedback Level [0.0..1.0]:** set the feedback level of the output signal back to the input. The feedback level changes the number of repetitions (echoes) of the effect.
- **Feedback HF Dump [20..16000 Hz]:** the feedback line is processed by a lowpass filter to emulate the absorption of reflecting materials, which higher for the high frequencies.
- **Panpot [-64..0..64]:** to set the balance between L&R channels of the output signal of each delay line. -1.0 means full to the Left channel, 0

means equally balanced between L&R channels, and +1.0 full to the R channel.

- **Level [0.0..1.0]**: sets the level of the output signal from each Delay unit.

Reverb 1 and Reverb 2

There are two reverb effects in DSP-AudioEditor. They differ from one each other for their internal algorithm. The first one is more intense, the second is lighter. Depending on the audio file that you are working with, one could be more suitable than the other.



- **L&R Gain Sliders**: on the left there is a section with two gain sliders, one for the Left, one for Right channel. Use these sliders to control the gain of the input signal. Peak and RMS Vu-Meters show the input L&R signal levels.
- **Efx Balance [0.0...1.0]**: to set the ratio between the processed and the input audio signals.
- **Time [0.0..1.0]**: to set the reverb time. This value is expressed as the ratio between the minimum value (time = 0) and the higher possible value (time=1.0).
- **Absorption [0.0...1.0]**: to set the factor which emulates the reflecting properties of the walls in function of the frequency spectrum. Lower values correspond to materials having an higher absorption at high frequencies.
- **PreDelay [0...100 msec]**: to set the time of the delay line on input to the reverb effect. It simulates the time that it takes the first echo to be heard due to the distance of the reflecting surface as it happens, as an example, when playing in a church or a very large hall.

Parametric EQ

The Parametric EQ of DSP-AudioEditor has six full parametric filters.

It is possible to set the frequency, gain, and bandwidth (Q) of each filter. Since all the filters have a full excursion on the frequency range, it is possible to overlap two or more filters, to let them to work on the same frequency range.

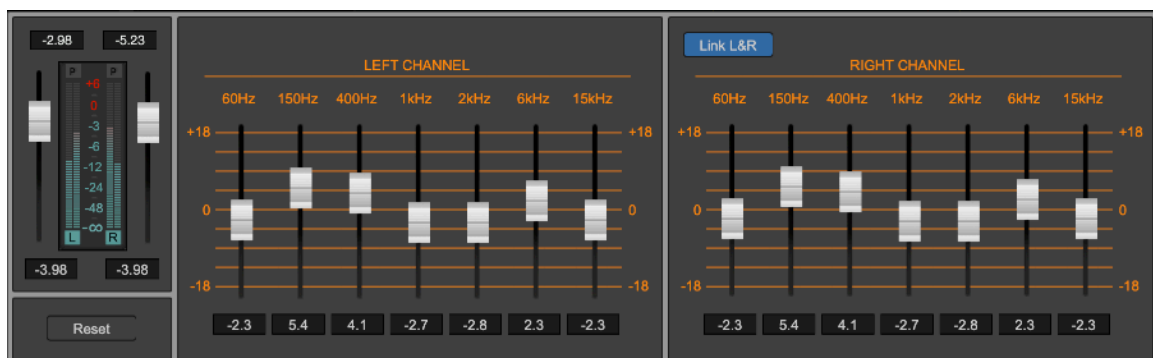


- **L&R Gain Sliders:** on the left there is a section with two gain sliders, one for the Left, one for Right channel. Use these sliders to control the gain of the input signal. Peak and RMS Vu-Meters show the input L&R signal levels.
- **Reset:** to reset each parametric filter to the default values.

For each parametric filter:

- **Gain [-18..+18 dB]:** to set the gain of the parametric filter.
- **Freq [20..16000 Hz]:** to set the central frequency of the frequency band.
- **Q [0.0..5.0]:** it controls the width of the frequency response curve of the filter. Higher values correspond steeper curves, ie a more selective filter.

Graphic EQ



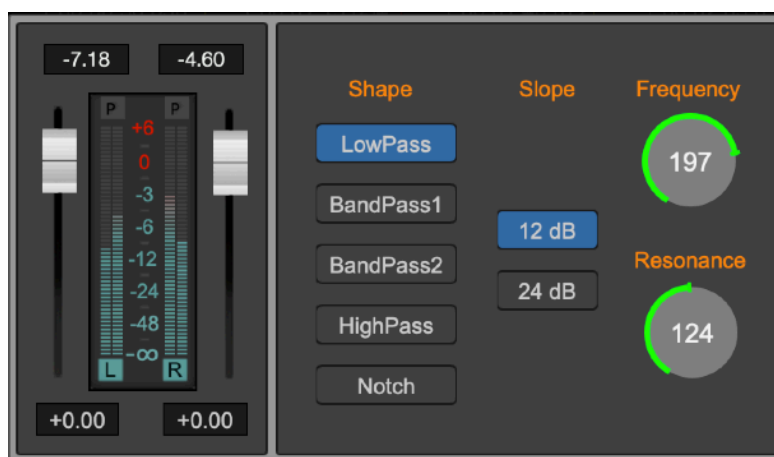
The graphic equalizer of DSP-AudioEditor built-in uses a bank of seven filters to modify the frequency contents of the audio signal. Each filter is centered around a different frequency according to a logarithmic scale. It is possible to set the gain of each filter in the ± 18 dB range.

- **L&R Gain Sliders:** on the left there is a section with two gain sliders, one for the Left, one for Right channel. Use these sliders to control the gain of the input signal. Peak and RMS Vu-Meters show the input L&R signal levels.

- **Reset:** to reset all gains to 0 dB.
- **Gain [-18...+18 dB]:** to set the gain of each frequency band. It is also possible to enter the numerical value clicking on the control below the slider.
- **Link L&R:** when turned On, setting a gain value for one channel, will set the same value for the other channel as well.

Resonant LowPass, BandPass, HighPass and BandReject Analog Filter emulations

Among the effects of DSP-AudioEditor, there are the filters that simulate the resonant analog filters, that became famous in the music world during the '70s, in vintage analog synthesizers. The following analog resonant filters are available: LowPass, HighPass, BandPass and Notch.



Each filter uses its own interface that is however very similar to the other. You can set the cutoff frequency, the resonance at cutoff frequency – called Resonance for LowPass and HighPass, or Q in the other cases – and the slope of the curve, measured in -dB/octave. Higher slopes bring to more selective filters, that are more resonant and that cut frequencies with steeper curves.

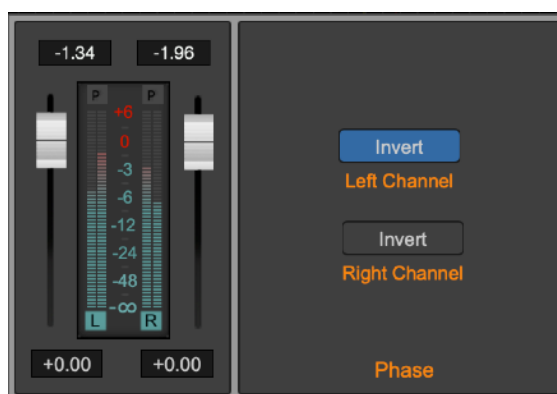
- **L&R Gain Sliders:** on the left there is a section with two gain sliders, one for the Left, one for Right channel. Use these sliders to control the gain of the input signal. Peak and RMS Vu-Meters show the input L&R signal levels.
- **Shape:** to choose among one of the following:
 - **Resonant LowPass:** it cuts all frequencies above the cutoff frequency. Slope can be set to -12 or -24 dB/Octave.
 - **Resonant BandPass 1:** it cuts the frequencies outside the band centered around the cutoff frequency.
 - **Resonant BandPass 2:** similar to BandPass 1, but a lot more resonant.
 - **Resonant HighPass:** it cuts all frequencies below the cutoff frequency. Slope can be set to -12 or -24 dB/Octave.
 - **BandReject:** it cuts the frequencies in the band just around the cutoff frequency. It can be very selective with high values of Q, and is very

useful when you want to remove from the audio signal a specific frequency, such as the power line noise at 50 or 60 Hz.

- **Slope:** to set the slope (-12 or -24 dB) of the filter.
- **Frequency [20..16000 Hz]:** to set the value of cutoff frequency.
- **Resonance/Q:** to set the resonance value or the filter bandwidth. Higher resonance values produce higher gains around the frequency of cutoff that may cause the auto-oscillation of the filter, exactly as it happens working with analog filters.

The Gainer and Phase Inverter

This is a very basic and very useful built-in plug-in. It applies a Gain with the option to invert independently the phase of the Left and Right channels.



External Audio Send&Return

This is a very special plug-in which sends the audio signal to an audio output device and which gets back the audio signal from an audio input.



It is an invaluable tool which is able to process the audio by using an external hardware audio device. Of course, it works in real time only - for this it is not available among the function for off-line processing.

Because DSP-AudioEditor has a built-in Output Recorder able to record to file its own outputs, using this output recorder and the External Audio Send&Return tool it is possible to process an audio file by an external hardware and, getting back the audio signal, to dump it into a new file. All this internally in DSP-AudioEditor.

This tool is useful if you have an external audio device having several I/O ports. If this the case, just select the output device audio ports to use as audio output to the external processor for the AUDIO SEND. Then, select the input ports of the input device to use to get the audio signal back after external processing on the AUDIO RETURN.

By default, DSP-AudioEditor mutes the audio returns to avoid unwanted audio feedbacks. Be careful turning the MUTE OFF.

NOTE: it is highly recommended to use the SAME HARDWARE audio device for both AUDIO SEND and RETURN to avoid any gap on the audio stream. These short audio gaps can be caused by a missed hardware audio synchronisation between Input and Output audio streams. This because if there is not a direct sync between different audio I/O devices, the 44.1 kHz of one device could be a little different respect the 44.1 kHz of another hardware device, causing occasional audio gaps on audio stream.

The Input/Output Audio Recorder

The input/output Audio Recorder of DSP-AudioEditor is able to record to file the audio on input to the Mac as well as the audio on output from the applications itself.

DSP-AudioEditor always creates an Audio Recorder at startup, and there is always only one Audio Recorder.

To show the Audio Recorder, use the command **Show Input/Output Recorder (or ⌘⌘R)** from the top menu bar of the application:

DSP-AudioEditor will show the Audio Recorder floating window:



The Audio Recorder interface consists of:

- Gain sliders and Vu-Meters



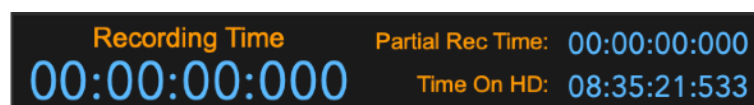
these are the same elements also used by the channel strip.

- The Transport Bar



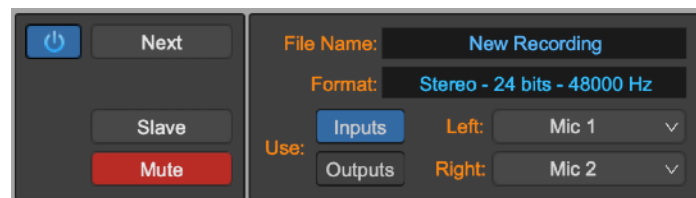
- **REC button**, to put a new recording on standby.
- **PLAY/STOP button**, to start and stop the new recording.
- **New Marker button**, to add a generic marker at the current recording position.

- Time Display

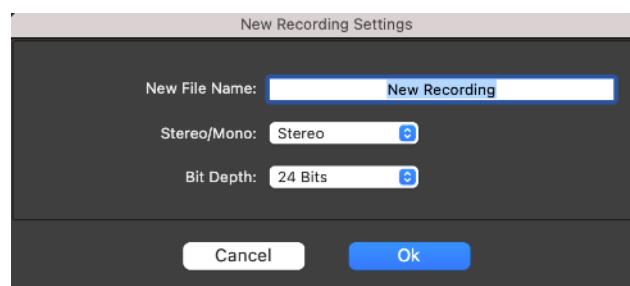


- **Current Recording Time:** it shows the time elapsed since the start of the recording session.
- **Partial Rec Time,** it shows the time elapsed since the last click on New Marker button, ie the position of the generic marker added last.
- **Time On HD:** the free space on the HD used for recording.

- **Other controls**



- **File Name and Format:** they show the current file name and file format. Clicking on it, DSP-AudioEditor shows a dialog to enter a new file name and the audio file format to use:



It is possible to set if the new recording must be a Mono or Stereo file and, if Mono, if the audio recorder must mix the input (or output) L&R channels, if it must use only the L or only the R channel.

It is possible to set the Bit Depth (8, 16, 24, 32 and 32 floats) as well.

- **Use Inputs/Outputs:** click on Inputs button to record the audio on input to the current audio device used by DSP-AudioEditor (refer to the chapter about the Audio Settings to know how to set this current audio device). Click on the Outputs button to record the audio on output from DSP-AudioEditor.
- **Left/Right popup menus:** if the audio recorder is recording the inputs, these popup menus show the list of the input audio ports to the current audio device used by DSP-AudioEditor. If the audio recording is recording the output from DSP-AudioEditor, these menus do not allow other choices other than Out L and Out R channels.
- **On/Off button:** to turn On/Off the audio recorder
- **Mute button:** to mute/un-mute the output of the audio recorder.
- **Next button:** to switch the file to the “next” on the fly during recording. Clicking on this button, DSP-AudioEditor closes the current recording, automatically switching to another file, naming it with a numerical extension to distinguish it from the closed recording. Then, DSP-

AudioEditor continues to record on this new file. DSP-AudioEditor adds the previous recording to the worksheet as well.

- **Slave button:** if ON, the audio recorder transport bar is slave to the transport bar of the Editor window. If the audio recorder is slave, clicking on the play/stop button on the transport bar of the main Editor window will control also the play/stop button of the audio recorder. In any case, to start/stop a new recording, the audio recorder must already be in stand-by, ie its Rec button must already be turned on, otherwise nothing happens.

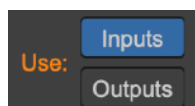
NOTE: turning the Slave button ON, it is possible to dump to file in real time the output of the audio file on the Editor.

NOTE: At startup, by default the audio recorder is muted to avoid unwanted audio feedbacks between inputs and outputs.

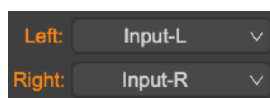
How to make a new recording

This is the step-by-step procedure to make a new recording:

- 1) Configure DSP-AudioEditor to use the wanted audio device using the **Options->AudioSettings...** menu command from the top application menu bar, as described in the relevant chapter in this manual.
- 2) Make the Input/Output audio recorder visible using the **Windows->Show Input/Output Recorder...** menu command from the top application menu bar or clicking on the recorder button on top of the Editor window.
- 3) **To make a recording from audio Inputs:**
 - I. **Click on the Inputs button** to record the audio inputs to DSP-AudioEditor

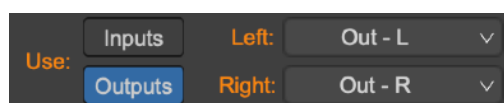


- II. **Use the popup menu controls on the Input Channels view to set the input ports for the L&R input channels:**



The VU-Meters of audio Recorder will show the input signal levels.

- 4) **To make a recording from DSP-AudioEditor Outputs:**
 - I. **Click on the Outputs button** to record the audio outputs of DSP-AudioEditor



The VU-Meters of audio Recorder will show the input signal levels.

- 5) **Click on the File Name** on the Audio Recorder module to enter the file name of the new recording and the audio file format.

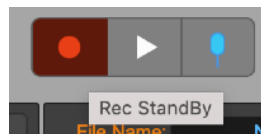


It is possible to set if the recording must be stereo or mono and, if mono, if the audio recorder must mix the input L&R channels, use only the L or only the R channel.

It is also possible to set the bit depth (8, 16, 24, 32 and 32 floats) of the new recording.

Click on the OK button to confirm.

- 6) **Click on the REC button to put the recording on STAND-BY.** DSP-AudioEditor waits to start to record (recording still does not start at this stage). The REC button will turn ON.



- 7) **To start recording:**

- I. **SLAVE mode OFF: click on the PLAY button on the Audio Recorder Transport Bar.** The counter shows the elapsed recording time.
- II. **SLAVE mode ON: click on the PLAY button on the main Editor window Transport Bar.** The counter shows the elapsed recording time.

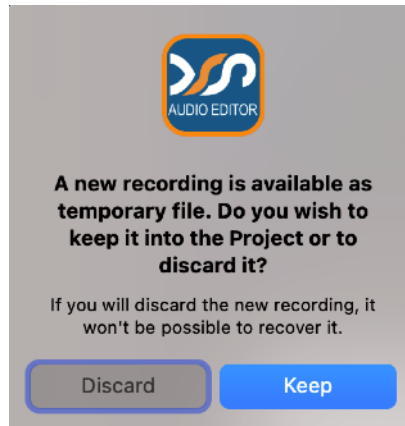
- 8) **To stop the recording:**

- I. **SLAVE mode OFF: click on the STOP button on the Audio Recorder Transport Bar.**
- II. **SLAVE mode ON: click on the STOP buttons on the Editor window Transport Bar or on the Audio Recorder Transport Bar.**

- 9) **The counter stops** and the PLAY button turns OFF

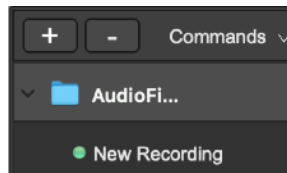
- 10) DSP-AudioEditor shows a dialog asking to keep or discard the new recording

If the recording is ok, click on KEEP to save the new recording.



WARNING: Discard will delete the new recording from the HD with no chance of recovery.

- 11) Then, DSP-AudioEditor adds the new recording to the Document Manager table adding it to the list of the audio files.



Double-click on the new file name row on the Document Manager table to open the new recording on the waveform Editor view.

NOTE: during recording, it is possible to click on the **New Marker button** on the Audio Recorder Transport Bar to add a generic marker at the current



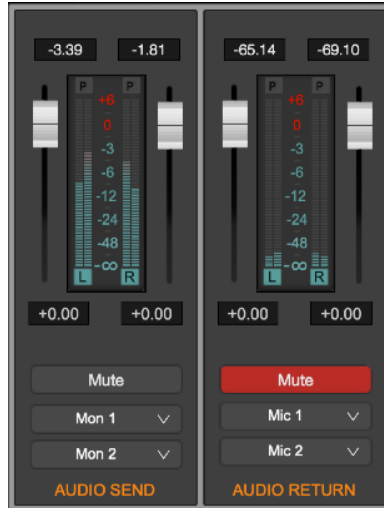
cursor position. DSP-AudioEditor will save these generic markers into the audio file at the moment of keeping the new recording into the worksheet.

Rendering to file the audio processed by an external hardware unit

To be able to use the audio Recorder as Slave to the transport bar of the waveform editor main view is an important feature of DSP-AudioEditor which allows to create a new file as result of the processing of an audio file through an external hardware effect unit.

To do it, open the Input/Output Recorder, set it to record the outputs. Then, click on the Slave button to set it slave to the waveform editor view transport bar. Set it on Stand-by clicking on the REC button.

Return to the editor window, and as Effect on insert on the current audio file choose the External I/O special plug-in of DSP-AudioEditor. Configure it to send and get audio to/from the external hardware effect.



Then, start to play the audio file on the waveform editor. The audio Recorder starts automatically to record because it is slave. Stop when finished, keep the new recording.

The new processed audio file will be listed on the Document manager table among the audio files.

NOTE: to work properly, due a restriction of the internal audio routing of DSP-AudioEditor, the sample rate of the current audio file to process must be the same of the sample rate of the current output audio device used by DSP-AudioEditor to play its output.

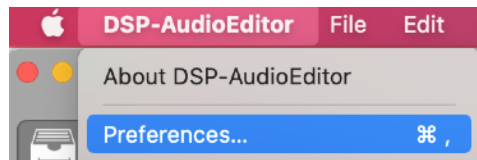
Preferences

Use the Preference dialog to set the default parameters used by DSP-AudioEditor.

The Preference dialog shows a multi-panel view which collects all the user configurable settings, divided into the following categories:

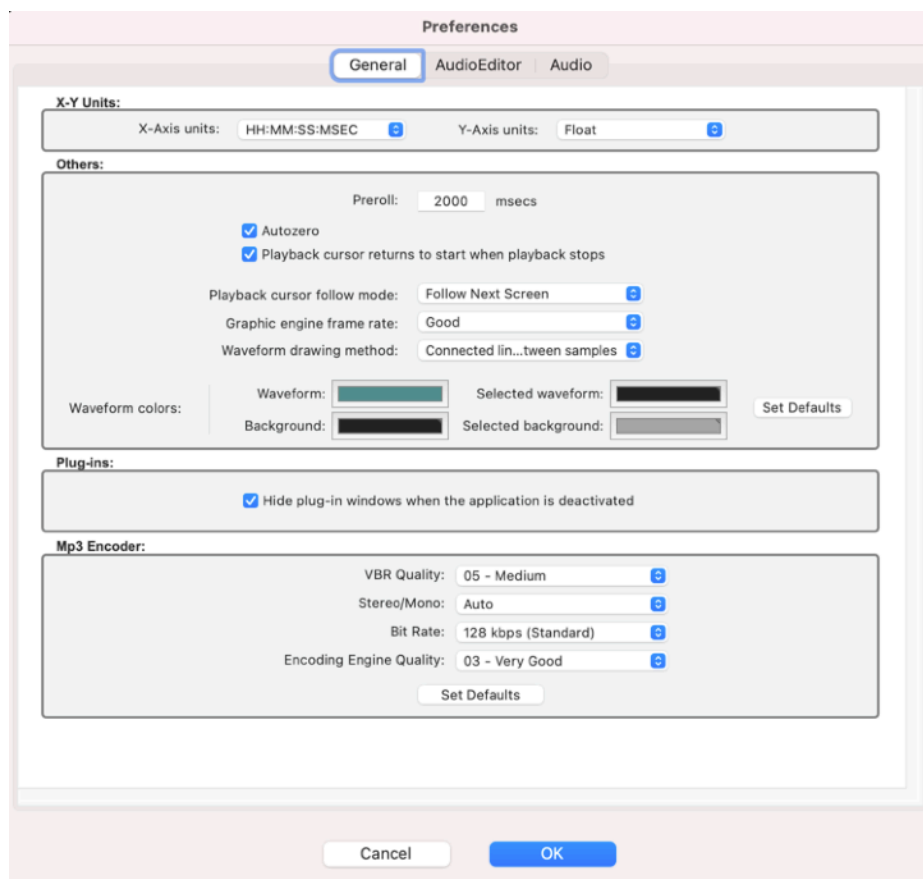
- **General:** general settings
- **AudioEditor:** Waveform Editor settings
- **Audio:** Audio Device settings

To open the Preferences dialog, use the **Preferences...** menu command from the application top menu bar:



In upper part there are the different tabs to choose the category of configurable settings to show on the lower part.

General Preferences



Selecting the General tab, the above view appears.

X-Y Units Options:

- **X-Axis Units:** to choose the default unit used by DSP-AudioEditor to represent the time axes on the Time Ruler and for displaying time values. Choose one of the possible options: hh:mm:sec:msec, Seconds, Samples, milliseconds or AudioCD.
- **Y-Axis Units:** the default unit used by DSP-AudioEditor to represent the level of the audio signal. Possible options are: floats (-1.0, 1.0), Decibels (dB).

Other Options:

- **AutoZero:** when this option is ON, releasing the mouse after moving a marker or setting the selection, DSP-AudioEditor automatically shifts the cursor point to the previous zero-crossing point. A zero-crossing point is the location in which the waveform intersects the zero level changing the sign. This option is very useful to place markers in positions that have the zero level in common.
- **Playback cursor returns to its starting position when the black stops:** if turned OFF, when DSP-AudioEditor stops to play, the playback cursor remains on the last position reached.
- **Waveform drawing method:** set the method used drawing the waveform. It has a big impact at high x-axis zoom-in levels, like 1:1 or more. Possible methods are: connected lines between samples, one dot for each sample, one vertical line for each sample.
- **Waveform Colours:** it allows you to assign the colours used to draw waveforms and backgrounds. Use the Set Default button to restore these settings to factory values.

Plug-ins:

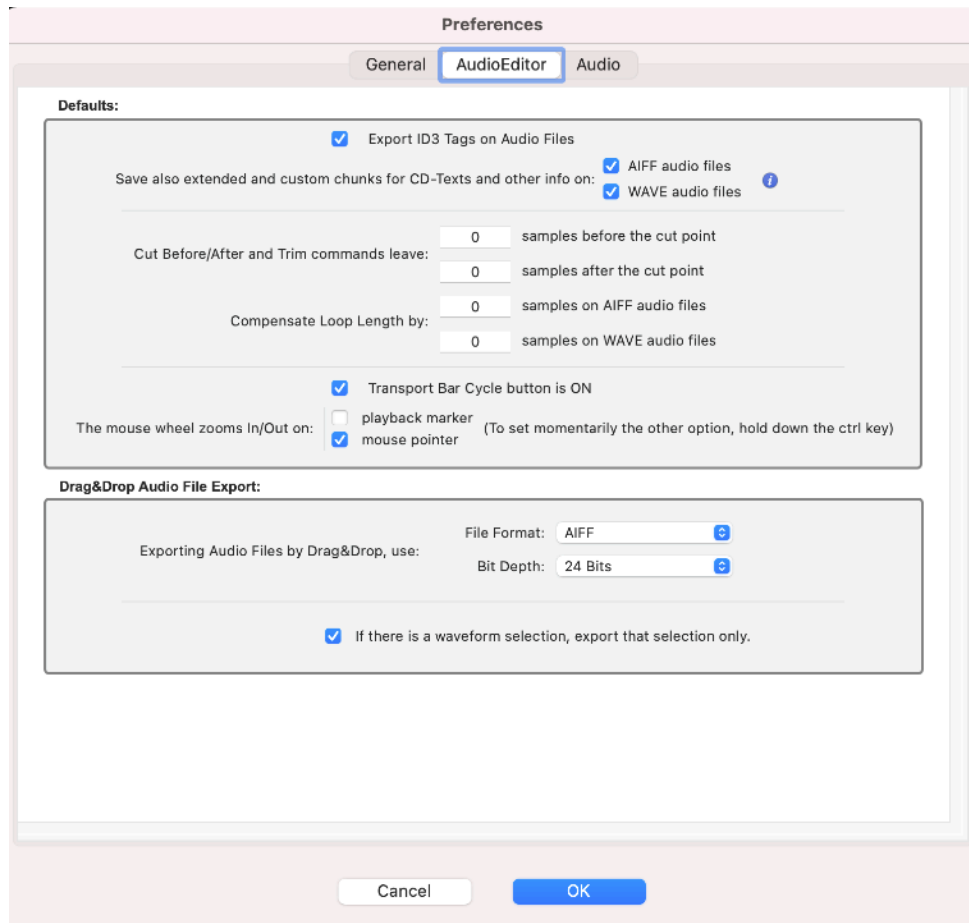
- **Hide plug-in window when the application is deactivated:** If OFF, when DSP-AudioEditor goes background, all plug-in windows remains visible on screen. Otherwise, they will disappear until DSP-AudioEditor becomes foreground again.
- **Restore Solo ON/OFF Status for each Channel Strip:** If OFF, when DSP-AudioEditor loads a Project document, it does not restore the Solo button status as it was when saving that Project.

Mp3 Encoder:

These are the advanced settings for the internal mp3 encoder. It is highly recommended to keep the default values as they are if not strictly necessary.

AudioEditor Preferences

Selecting the AudioEditor tab, this view appears:



Defaults:

- **Export ID3 Tags on Audio Files:** if turned On, DSP-AudioEditor will write also the ID3Tag chunk saving audio files. Be aware that not all applications support ID3Tag standard. If an external application or services has problems reading audio files generated by DSP-AudioEditor, turn OFF this option for wider compatibility.
- **Save also extended and custom chunks for CD-Texts...** Usually custom information and additional chunks for CD-Texts are well supported by other applications, but some hardware samplers are not able to read these chunks on AIFF and WAVE audio files, failing to read correctly also the loop information. Because, even if this option is turned OFF, DSP-AudioEditor always saves the loop information on AIFF and WAVE audio files, it is recommended to turn this option OFF when using DSP-AudioEditor to build sample libraries for software or hardware samplers.
- **Cut Before/After commands leave:** it allows you to set two values for the number of samples that DSP-AudioEditor will leave before and after the cut points, when it is applicable. It is very useful doing sample editing,

because several samplers need of some samples after the loop end to loop correctly.

- **Compensate Loop length by:** playing loops on different hardware or software samplers, a potential problem can arise when playing loops, because there is not a standard saying if the loop point must be considered part of the loop or not. It means that in some cases it is necessary to adjust of 1 sample the loop length to play it without any audio glitch. To know if this is necessary or not, create a sine wave audio file for testing, and place the loop points such that it plays without any glitch using DSP-AudioEditor. Use this option if playing the same audio file on the destination sampler there is a glitch at loop point to adjust the loop length during the import/export until the same file plays perfectly the same in both cases.
- **Transport Bar Cycle button is ON:** if this flag is ON, the Cycle button on the Transport bar will ON by default for new Editor windows.

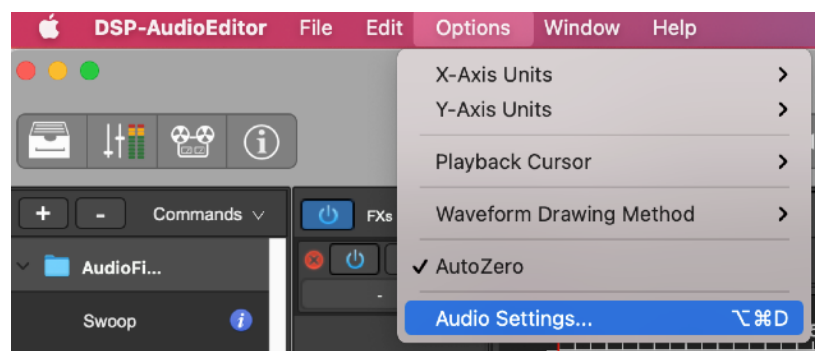
Drag&Drop Audio File Export:

- **Exporting Audio Files by Drag&Drop, use:** it allows to set the file format and bit depth of the destination audio files when exporting an audio file by Drag&Drop to the MacOS file system.
- **If there is a waveform selection, export that selection only:** If OFF, exporting by Drag&Drop to the MacOS finder will always export all the file, even if there is waveform selection.

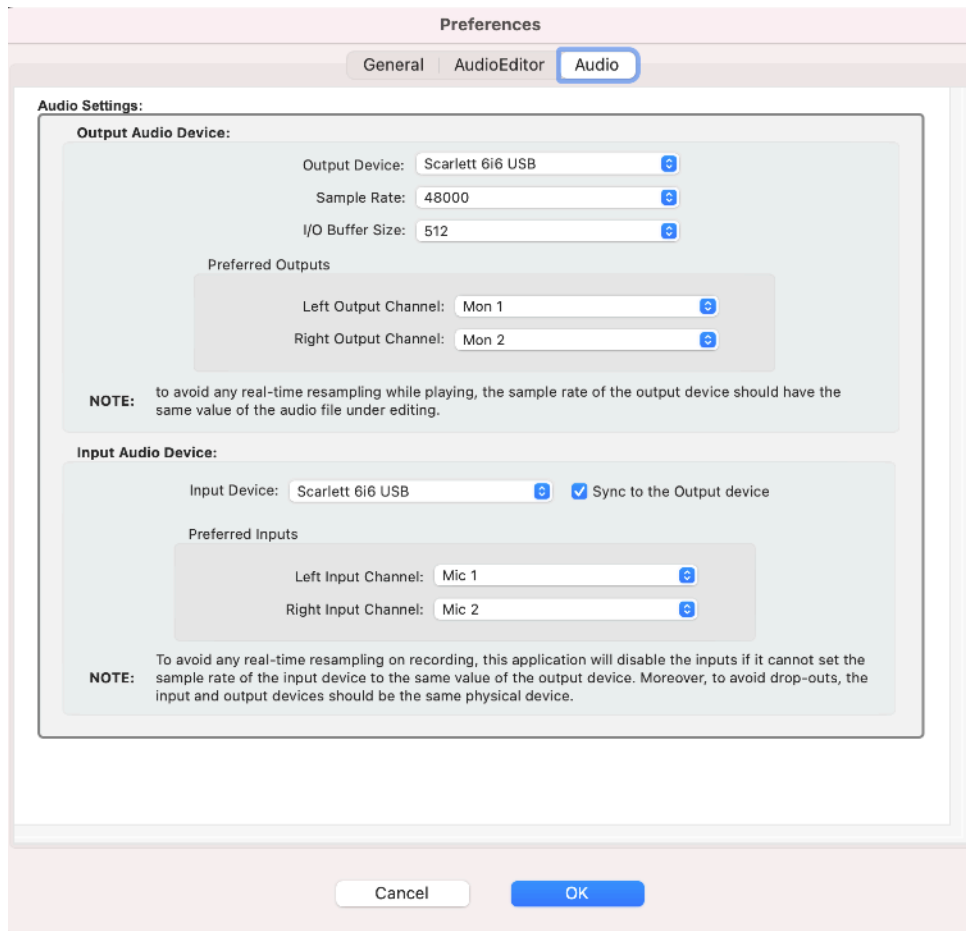
Audio Settings

Even if DSP-AudioEditor doesn't need any additional audio hardware to work properly, it supports and manages all audio devices which are compatible with CoreAudio of MacOS. To verify if an external audio device is compatible with MacOS and DSP-AudioEditor, run the Apple Audio MIDI Configuration utility which is into the Applications/Utility folder. If an audio device is visible among the list of available audio devices on the Audio Configuration window, then DSP-AudioEditor is surely be able to use it for audio I/O.

NOTE: To assign and configure the audio device used by DSP-AudioEditor for audio input/output, it is also possible to use the menu command on the application top menu bar **Options->AudioSettings...** menu command from the application top menu bar:



These are the options for Audio I/O settings:



- **Audio Settings:** in the upper part, two popup menus allow you to set which I/O device must be used by DSP-AudioEditor to play on output (output device) and to receive audio on input (input device).

Among the available devices, there is always a “System” device, which is the one assigned as default to MacOS for audio I/O. By default, DSP-AudioEditor starts using the System device.

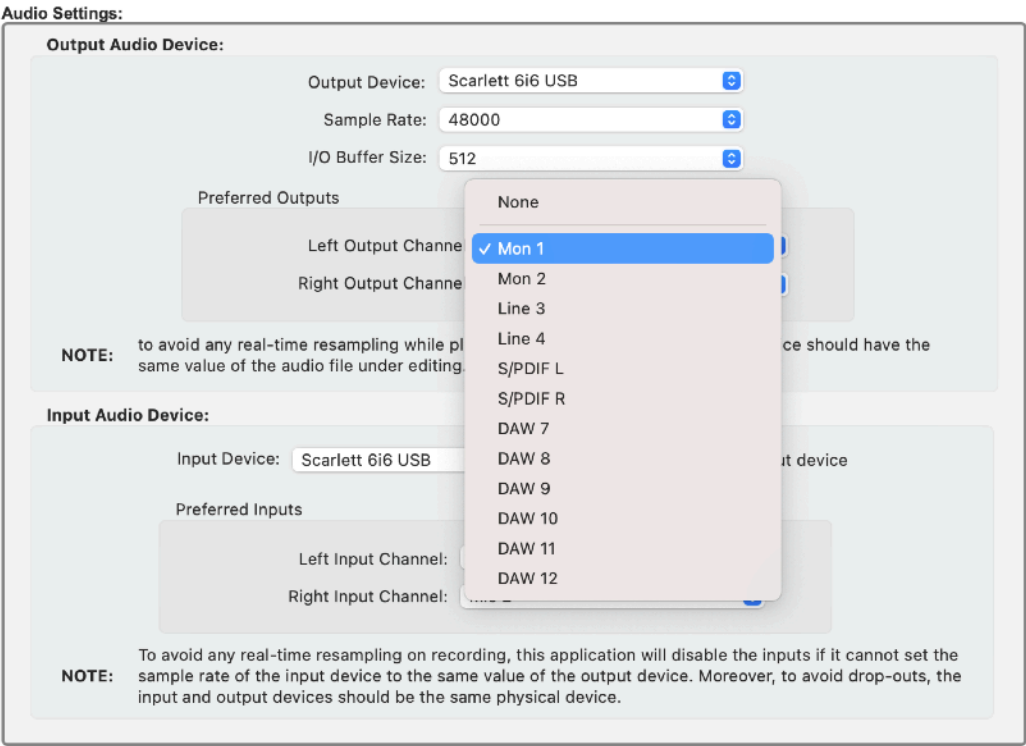
NOTE: it is highly recommended to set the same physical audio device for both inputs and outputs. Other settings may produce audio drop outs and clicks.

Turning ON the **Sync menu selections** check box, choosing an audio device for output and/or input, DSP-AudioEditor tries to do the same choice for the other device (input device when choosing the output, and the contrary) automatically. This with the purpose to assign the same physical hardware interface for both Inputs and Outputs. It is highly recommended to keep this option On.

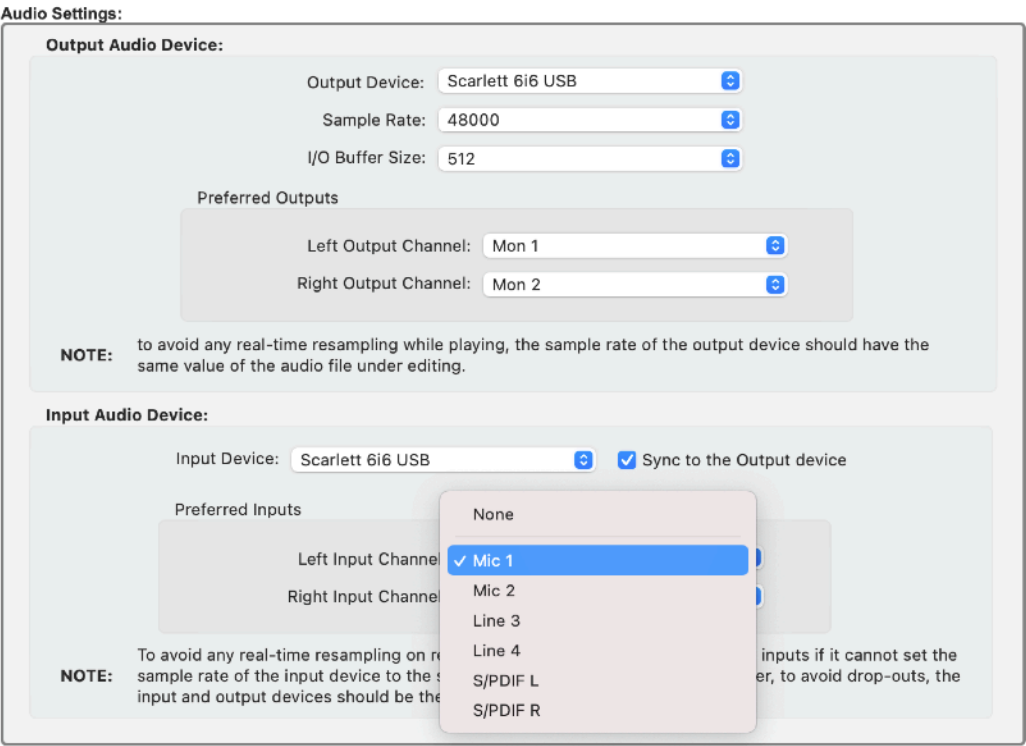
- **Preferred Inputs/Outputs:** these popup menus list the available audio input or output ports available on the current audio device (ie the audio device which is selected on the I/O Audio Device popup menu above).

These preferred Input and Output channels are the channels which DSP-AudioEditor uses by default as input ports on the audio recorder and which uses for playing its output.

As an example, choosing the Scarlett 6i6 as input and Output audio device, which is a multi-port audio device, this is what DSP-AudioEditor shows as a possible choice for the preferred output channels:

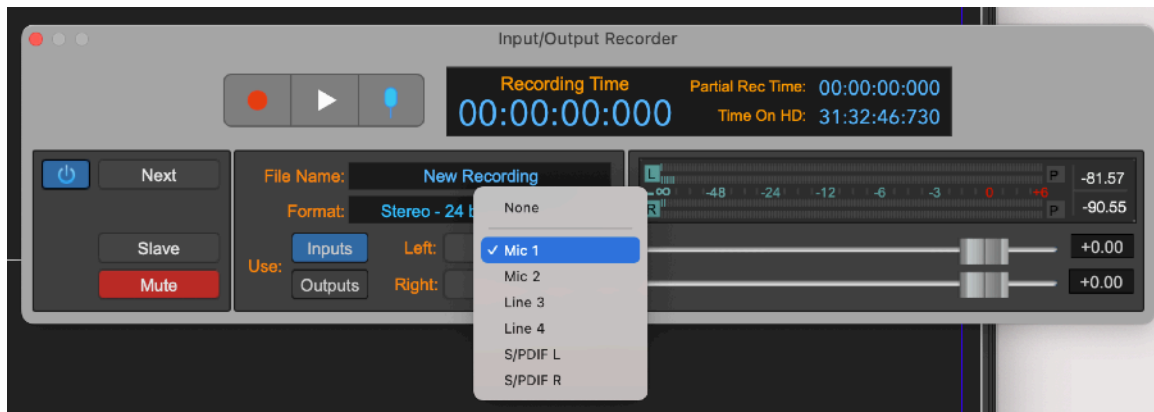


And this for the preferred input channels:



With the above settings, DSP-AudioEditor plays always using the Mon1 and Mon2 output audio ports of the Scarlett 6i6 audio device. Opening the Audio Recorder to record the Inputs, the Mic1 and Mic2 are the default input ports.

It is possible to set a different audio port for audio inputs:



For outputs, if it is necessary to use a different audio port for playing outputs, use again the Audio settings opening the Preferences dialog or from the Options->Audio Settings menu command.