


# Soundscape V7

**User Guide**

Soundscape. This is SSL.

Solid State Logic
SOUND || VISION

**Document History**

| | | |
|---|---|---|
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| January 2013 | 82BSSP01B | Initial Release for V7.0 |

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# 1. Introduction

Welcome to SSL Soundscape V7, the High Performance DAW from SSL and the perfect companion for your SSL MX4 or MadiXtreme audio card. Ultra reliable 128 channel simultaneous recording/playback, SSL plug-ins, 128 channel MADI I/O and advanced DAW editing and workflow paired with the pristine audio quality you expect from any SSL product.

Should you ever need assistance in setting up or using your sound card, Solid State Logic's website contains extensive FAQ section:

www.solidstatelogic.com/support/io.asp

*Please register your SSL audio card and Soundscape Software on our website. This will ensure that you receive notifications of future software and driver upgrades and other important information, and that your guarantee is registered. Registration will also make you eligible for technical support. Visit us at www.solidstatelogic.com.*

Solid State Logic is committed to the development and marketing of professional solutions for native PC and Mac based digital audio recording systems.

The Soundscape DAW and MX4/Mixpander cards, used in combination with our XLogic Alpha-Link audio converter products, provide a flexible, professional quality, high channel count audio solution for PC based audio recording, editing and mixing.

## How to use this manual

The SSL product range has been designed from the ground up to be easy to use. If you are familiar with the Windows environment, installing PCIe, and PCI cards and the basics of recording and playing back digital audio, you could probably just set the system up and feel comfortable running a session within an hour.



**If possible, please do not print this manual.**

However, Soundscape offers a wealth of powerful and helpful features that you will only discover quickly by reading this manual. It is therefore advisable, at some point, to read it from cover to cover. For example, Soundscape offers a wealth of precise Audio Editing Tools and the Mixer is fully configurable, and while you may find it simple to edit your recordings with simple tools you are familiar with from other DAW's and a few ready-made Mixers initially serve your needs, to really harness the power of Soundscape, read Chapter 4 of this manual as soon as you can.

Please make sure you understand the Master Clock and Sample Rate concepts and that you understand the software's hardware settings. It is also a good idea to have the system switched on while you read the manual, so that you can experiment with the features you read about.

We trust that you will soon feel confident creating and using your own Projects. However, even when it has become second nature, the comprehensive Table of Contents (located at the beginning) and the search function in your PDF reader software will provide convenient ways to check specific information whenever you need it.

## Supported hardware

Soundscape V7 supports the MX4, MadiXtreme and Mixpander audio cards. The information in this manual relates to these cards. Differences are pointed out where necessary.

Legacy Soundscape Systems (eg. the Soundscape 32 System) can be used in a limited way with Soundscape V7:

- MIDI sync device (MTC, MIDI Clock), or as Time Code Sync and 9 Pin (requires Sync Option) connected to a PCI Host IF card
- As an I/O Device when connected to a Mixpander card;

However, the Soundscape user guide does not go into too much detail about legacy systems. Please retain the legacy version user guide for when you plan to integrate a legacy SS system.

# Reading conventions

## Key commands and key combinations

Some Soundscape functions can be accessed through the use of computer keyboard keys or key combinations, as well as by using a mouse or other input device. In this manual computer keys will be shown in bold italic, with + sign indicating key combinations. For example, the letter E key would appear as ***E***, and 'Control E' would appear as ***Ctrl+E***.

## Menus

Where appropriate, to indicate a 'path' under one of the main menus, the following format will be used:

Menu: **Header** / **Submenu 1** / **Submenu 2** / **Submenu3** / **Item**.

## Screenshots

The appearance of the Soundscape software on your computer screen may be different from the screenshots in this manual.

## Labelling

As a general rule, **bold** type indicates that the text precisely reflects on-screen labelling.

## Internal References

References to elsewhere in this user guide are displayed in *italics*. 'Single quotation marks' are used when the text is already in italics. Internal references are also hyperlinked – clicking on them will navigate your pdf reader to that page.

# WEBSITE

The URL for the Solid State Logic website is: www.solidstatelogic.com

The SSL Support Website is: www.solidstatelogic.com/support



# SSL Soundscape – The SSL DAW

## High Performance SSL Audio Engine

SSL Soundscape is powered by SSL's MX4 technology, built from the legendary Soundscape DAW's heritage. The completely re-engineered Soundscape is reborn to be blazingly fast, ultra-reliable and completely optimised for modern Multi-Core Processor Architectures and Windows 7 (32-bit or 64-bit).

The brand new Soundscape Hybrid Core Audio Engine allows recording, editing, overdubbing and mixing of 128 Tracks on a stunningly relaxed PC, with many hundreds of zero latency DSP and native VST Plug-Ins. Unlike any other DAW Software on the market, Mixing and Monitoring can happen with DSP FX completely 'in the box' with superior pristine SSL Console Grade Sound Quality and with almost no latency (4 samples 'roundtrip').

## Advanced and collaborative Editing

Sophisticated editing tools and workflow strategies, extreme speed and performance and a clean and intuitive workspace for Audio Pros. Soundscape is made for speed and to keep the user in a creative flow.

SSL Soundscape uses standard local or networked Windows storage to record and playback uncompressed 16 Bit and 24-bit Soundscape Takes, Wave and Broadcast Wave Files. A Soundscape Project can also contain a mixture of file-formats and bit-depths avoiding the need for awkward file conversions. For collaborative projects, there's no need to copy files to colleagues' computers, Soundscape allows safe collaboration over a standard Gigabit Ethernet Network.

## Digital mixing, effects and processing

The SSL 'Console' inside Soundscape runs on the on-board DSP-powered mixing engine, providing immense audio processing capabilities. The Mixer's architecture is highly flexible, putting no limits on how the channels are structured.

The SSL Console EQ-Filters, Channel Dynamics, and the legendary Bus Compressor plug-ins offer the highest quality processing you can find in digital audio and provide console grade processing for that 'hit record' sound.

The SSL Audio Toolbox provides essential building blocks, with multi-function dynamics processors (gate, expansion, compression, and limiting), delay based effects (multitap delay, chorus, flanger) and dither. Optional effects and processing plug-ins are also available from other world renowned developers.

## DSP-based Hardware Processing

PC-based mixers suffer from a certain amount of processing delay, ('latency'). This may be very small on a well configured PC, but gets worse as native effects and processors are added into the signal path, so much so it can be impossible to play an instrument and monitor the output in real-time through a software mixer with a few plug-ins. This is why most native MIDI+Audio sequencers now include direct Monitoring (without FX) and a 'plug-in delay compensation' feature, which only solves the problem in mixing situations, but is unusable while recording, tracking and monitoring.

In contrast, the DSP-powered plug-ins offer a level of performance on a par with high-end audio hardware in terms of sound quality and comparable to a hardware mixing console in terms of latency (...or absence thereof!). This is a major advantage when recording live vocals or instruments. DSP effect plug-ins can be inserted at any point in the signal path and the wet signal can be monitored in real-time (ie. without any annoying processing delay) while recording the dry or wet signal, or both.

## Native effects and DSP processing plug-ins

While SSL format DSP-powered effects and processing plug-ins provide a unique combination of superior sonic quality, negligible latency and rock-solid reliability, we appreciate that native processing has a part to play. Soundscape supports the VST format, running on the host CPU, and VST FX Plug-Ins can be inserted directly in the SSL Mixer, seamlessly, including full support of dynamic automation of all knobs. This is useful in situations where latency is not an issue (eg., during mixing and mastering) and allows access to hundreds (if not thousands) of plug-ins.

*Note that MadiXtreme cards have no on-board processing. The features mentioned above, therefore, do not apply.*

*A blank page, nothing to read here...*

# 2. SSL Soundscape V7 Software Installation

## Important Preparations

*Before you proceed installing Soundscape, please ensure that you have successfully completed the Hardware Installation, as described in the MX4 Installation Manual, otherwise you will not be able to successfully install or work with the Soundscape or Mixer application.*

The files required for the following Installation can either be found on the CD which came with your audio card (SSL MX4, MadiXtreme 64, or MadiExtreme 128) or can be downloaded from the SSL website.

The audio card CD comes with an Install Menu Application (MX4_start.exe, for example), which allows easy Installation of all software components.

In order to avoid any complications during software installation please follow these steps in the order indicated:

1 Make sure you have installed the audio card and its drivers properly. If you have an internet connection, please download and use the most recent version, available from the SSL website.

2 Run the Soundscape V7 Combo Installer and as well as any optional extras – Pro-Convert V7 for Soundscape and SSL Console Control for Soundscape.

3 Start the SSL Soundscape V7 Software and follow the Instructions of the Unit Configurator. Make sure the Software runs properly.

4 Enter the Mix Password you find on your audio card Registration card, under **Options** / **Passwords** / **V2.xx Mixer** starting with the letter M.

*Note that the MX4 comes with plug-in passwords. If you have no plug-ins to install, please ignore instructions 5 to 8 below:*

5 Enter the SSL Console EQ Password you find on your audio card's registration card (SEQ), under **Options** / **Passwords** / **SSL Console EQ** starting with the letter M.

6 Enter the SSL Console Dynamics Password you find on your audio card's registration card (SCD), under **Options** / **Passwords** / **SSL Console Dynamics** starting with the letter M.

7 Enter the SSL Bus Compressor Password you find on your audio card's registration card (SBC), under **Options** / **Passwords** / **SSL Console Bus Compressor** starting with the letter M.

8 Enter the SSL Audio Toolbox Password you find on your audio card's registration card (ATB), under **Options** / **Passwords** / **Audio Toolbox (Partx)** starting with the letter M. You only have to enter the password once.

9 Enter the SSL Soundscape Password you find on your Soundscape webshop receipt or in your 'MySSL' account online (SS6), under **Options** / **Passwords** / **Soundscape Version 6** starting with the letter M.

10 Enter the Time Module Password you find on your Soundscape V7 webshop receipt or in your 'MySSL' account online (XTM), under **Options** / **Passwords** / **XPro Time Module** starting with the letter M.

11 Connect your Converter or MADI Console and set Clocking and MADI Mode under **Settings** / **Master Clock** and **Settings** / **MADI** / **MADI Mode**.

12 Ensure that the software runs properly. Under **Settings** / **Save Settings** you can make your changes permanent.

All the steps above are described in greater detail on the following pages.

## Compatible SSL Audio Cards

SSL Soundscape has been designed to take advantage of SSL's DSP and Audio Core Technology, available on the MX4 MadiXtreme 64 and MadiXtreme 128 cards, each of which have been optimised for modern Multi-CPU Processors and Windows 7.

For best results, please only use the following SSL Audio Cards:

### MX4

MX4 PCIe card features two MADI inputs and two MADI outputs, providing 128 simultaneous inputs and outputs at up to 48kHz, or 64 simultaneous inputs and outputs at up to 96kHz. And a set of SSL optimised hardware DSPs that allows it to perform the most demanding mixing tasks.

*Note will need MX4 Firmware version 19 or higher. Please go to the MX4 Support page on the SSL website if you need to run a update.*

### MadiXtreme 64 and 128

Featuring one or 2 MADI heads for 64 or 128 simultaneous inputs and outputs, MadiXtreme PCIe cards offer a cost effective option for use with external mixing.

### Mixpander

The SSL Mixpander PCI card can accept up to 64 simultaneous I/O at 48Khz via its expansion port when connected to an SSL Alpha-Link interface. Two Mixpander Cards can be used simultaneously.

*NOTE: Soundscape can also integrate legacy Soundscape Systems (REd 16-32, SS16-32) as I/O (when connected to a Mixpander Card ) and/or Sync Unit (connected to a PCI Host Card). For more information about the integration of legacy Soundscape Systems into a Soundscape setup, please visit our FAQ Section online, where you will find useful information and application notes:*

www.solidstatelogic.com/music/soundscape/faq.asp

## System Requirements

Soundscape V7 software, and the MX4 and MadiXtreme 64/128 PCIe cards, are compatible with the following operating systems and driver protocols:

| Platform | Operating System | Driver Protocols |
|---|---|---|
| PC with 32-bit Windows | Windows XP SP2 or later | ASIO 2.x (32-bit), WDM, MME, GSIF2, DWave, SSL Soundscape V7.0 |
| | Windows Vista SP1 or later | |
| | Windows 7 or later | |
| PC with 64-bit Windows | Windows Vista SP1 or later | ASIO 32-bit, WDM, ASIO 64-bit, SSL Soundscape V7.0 |
| | Windows 7 or later | |

**MAC OSX:** MX4 can run under OSX Leopard V10.5.8 or greater, on OS X Snow Leopard V10.6.1 and on OS X Lion V10.7.2 or later (32-bit and 64-bit) by using MadiXtreme Core Audio Divers. Under MAC OS the MX4 Card works as a MadiXtreme 128. The Mixer Software and DSP Plug-Ins do however not work with MAC OS.

The SSL MX4 and MadiXtreme Card Family comes with low-latency 32-bit MME drivers, WDM drivers, ASIO-2 drivers, DWave drivers and GSIF drivers for Windows XP, Vista 32 and Win 7 32, as well as ASIO 32/64 and WDM 64 Drivers for Vista and Win 7 64-bit Versions. It can be used with any PC based MIDI & Audio sequencer, recording and editing software or other audio applications. The SSL Soundscape Drivers are truly multi client, allowing you to share your SSL audio hardware between several applications that use different driver models.

The SSL Mixpander PCI card and Soundscape V7 software are compatible with the following operating systems and driver protocols:

| Platform | Operating System | Driver Protocols |
|---|---|---|
| PC with 32-bit Windows | Windows XP SP2 or later | ASIO 2.x (32-bit), WDM, MME, GSIF2, DWave, SSL Soundscape V7.0 |
| | Windows Vista SP1 or later | |
| | Windows 7 or later | |
| PC with 64-bit Windows | Windows Vista SP1 or later | ASIO 32-bit, WDM, ASIO 64-bit, SSL Soundscape V7.0 |
| | Windows 7 or later | |

The SSL Mixpander comes with Low-latency MME drivers, WDM drivers, ASIO-2 drivers, DWave drivers and GSIF drivers for Windows XP and Vista. It can be used with any PC based MIDI & Audio sequencer, recording and editing software or other audio applications. The SSL Soundscape Drivers are truly multiclient, allowing you to share your SSL audio hardware between several applications that use different driver models.

On Windows Vista and Windows 7, a maximum of 2 GB RAM is recommended, otherwise Mixpander may not work properly.

## System Recommendations

Soundscape has been designed for the latest PC Core Technology (CPUs, Chipsets, HDDs) and Windows 7 Operating Systems.

As technology is constantly evolving, for up to date System Recommendations and a list of Soundscape certified Turnkey Systems from 3rd Party vendors, please visit our System Spec page at:

www.solidstatelogic.com/music/soundscape/specs.asp

# Installing from Web Download

Start by downloading the Mixer/Soundscape V7 Combo Installer from the SSL website:

www.solidstatelogic.com/music/soundscape/downloads.asp

When you unzip the downloaded file, you will find four folders within it.

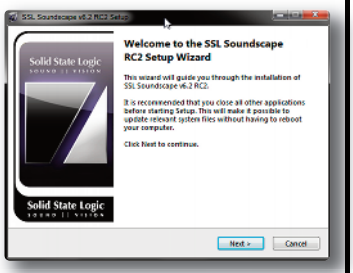
1. The SSL Soundscape family driver; This is a driver which allows all of the relevant SSL sound cards to be run on the computer. Note that your SSL sound card comes with a CD which also contains this driver. Installation of the driver is described in the documentation included with each product.
2. The SSL Soundscape & Mixer; This is the main installer for Soundscape and is described below in detail.
3. Console Manager; This installer is included for legacy purposes as the software has been replaced by SSL Console Control.
4. Resources; This folder includes all user documentation referenced by Soundscape's **Help** menu.

## Main Installation Procedure

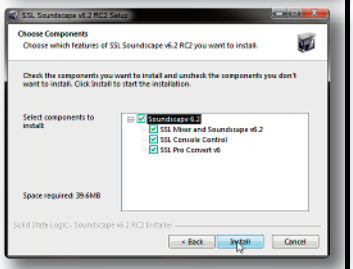


Simply unpack the download into a folder and start **SSL Soundscape Setup v7** combo installer wizard. (Depending on the Windows version, you may be prompted with some dialogue boxes to confirm that you really, really want to run this Installer)

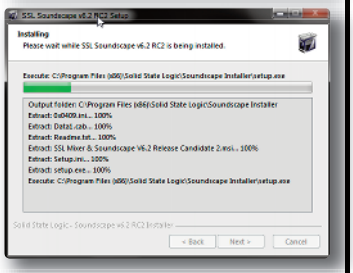
Click **Next**.




In the next screen, choose the components you wish to install – Soundscape and its Mixer, along with optional extras Console Control and Pro Convert – and click **Next**.




The combo installation will now begin, as indicated by the progress screen which is displayed.



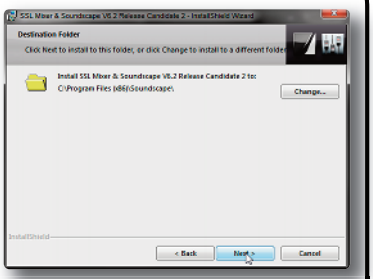

During the combo installation, the main Soundscape setup dialogue will open in a new window – click **Next** to start the Soundscape installation process.

*Note that there may be a short pause before the buttons become available.*



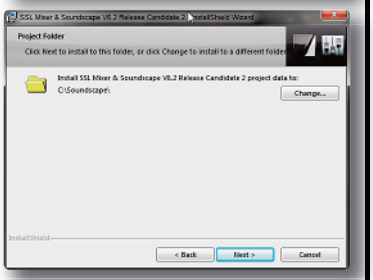
Please note that although you can select a different installation folder than the default offered, the SSL Mixer & Soundscape user manuals and support procedures will always refer to the default installation folders. We therefore recommend using the default installation locations.

Click **Next**.



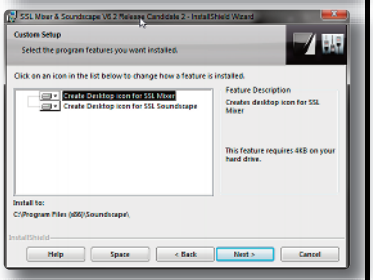
You can also select different a Projects Data Folder. Again, however, we recommend using the default location.

Click **Next**.

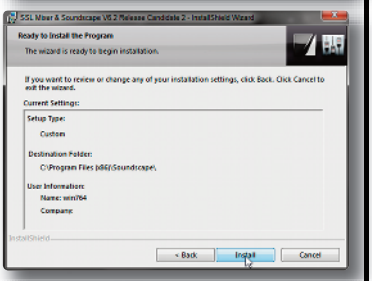


Once you have selected your installation and project data folders, you may select whether Soundscape and Mixer icons should be placed on the desktop.

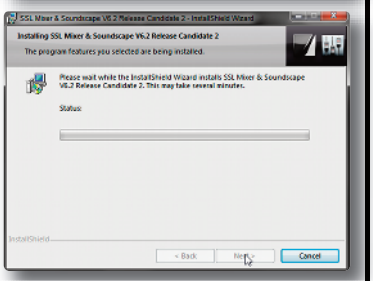
Click **Next**.



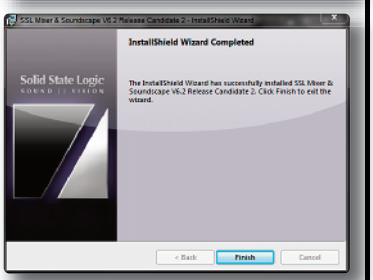
Now you are ready to install some very fine SSL Software... Click **Install** to complete this part of the installation.



A progress bar will allow you to monitor the progress of the installation. Click **Cancel** at any time if, for any reason, you need to abort the installation.



Once the Soundscape Installer has successfully finished, you can click **Finish** to close the installer and return to the combo installer progress bar.



Once the Combo Installer has successfully finished, you can click **Finish** to complete the last part of the installation process.




*Note: The SSL Mixer / Soundscape Combo Installer will also install the latest Version of the SSL Console Bundle and Audio Toolbox.*

*Note also that if your computer does not have Miscrosoft .Net installed, an installer for it will commence automatically.*

# Installing DSP and VST Plug-Ins

## SSL Soundscape format DSP plug-ins

If you have SSL Soundscape plug-ins to install which were not included in the main installation, you will find note on the installation in the documentation which came with them.

First, close all the Soundscape or Mixer Applications, double click on the plug-in's installer application icon and follow the instructions.

If you have installed Soundscape in its default folder, the installation will be straight forward; If Soundscape is installed in a location other than the default folders you might need to point the Installer to your Soundscape install location.

After the plug-ins have been correctly installed, launch Soundscape and enter the plug-In password you have received by email into the appropriate line in the **Options** Menu. Authorising plug-ins is explained in detail in the '*Options Menu*' section of Chapter 5.

*Please note that when working with Windows Vista or Windows 7, you may have to run the installer in compatibility mode for Windows XP (right-click **Installer Program / Properties / Compatibility Tab**).*

*Note that DSP plug-ins are not available with a MadiXtreme.*

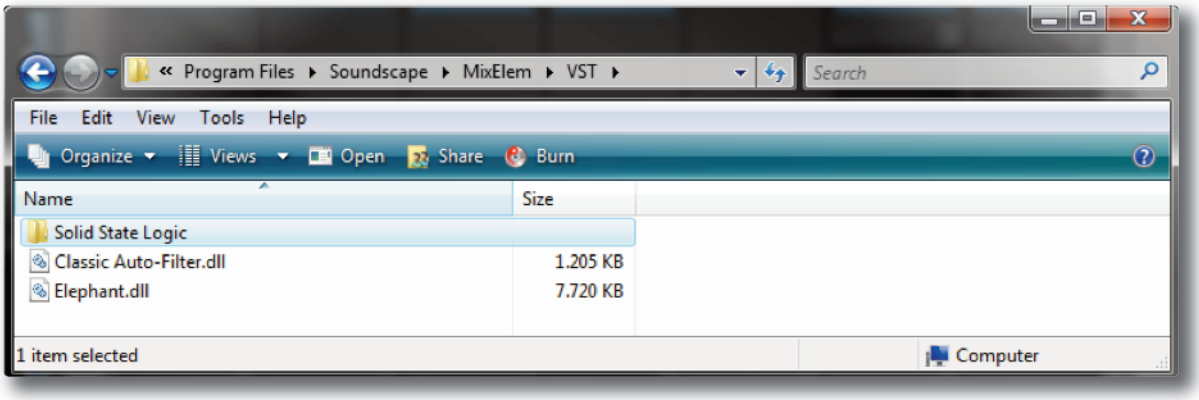
## VST format plug-ins

The process of installing VST format plug-ins may vary depending on the developer's installation process. However, many VST Manufacturers install the VST.DLL files by default in: **C:\Program Files\Steinberg\VstPlugins**.

For Soundscape to be able to work with installed VST plug-ins, their DLL files must be present in SSL Soundscape's VST folder, which by default is:

**C:\Program Files\Soundscape\MixElem\VST**

If it is not possible to re-direct the plug-in installation to Soundscape's VST folder you will need to manually COPY the DDL files to Soundscape's VST folder.




At startup, Soundscape scans the VST folder and subfolders (Solid State Logic in the example above) and a small window appears in front of the splash screen where the plug-in name is displayed while loaded.

*Note that the Program Folder for 32-bit Apps on Win 7-64 (or Vista 64) is called **Program Files (x86)***

# 3. Soundscape Quick Start Guide

The purpose of this chapter is just to get you going with Soundscape. You will learn how to configure your system and create your first project, then we will introduce you to a handful of Soundscape-specific terms and workflows.

Experienced Soundscape operators will learn how to migrate their Projects to V7 at the end of this chapter.

*Please Note: This chapter does not explain functionality in detail. See Chapter 4 for a details of the functions described, and then Chapter 5 to really understand the unique Soundscape concepts and how they help you to speed up your audio life function by function.*

## SSL Mixer or SSL Soundscape?

The SSL Mixer/Soundscape installer includes two applications, both of which share identical mixing functions, run on the same hardware and (in many applications) do the same Mixing job; However, they are developed to perform different tasks:

### SSL Mixer

The Mixer's main purpose is to work with other audio software such as Cubase/Nuendo, Sonar, Reason, etc, taking care of routing and audio processing (before or after the DAW's software input) and performing mixing and monitoring tasks in 'DSP Realtime' (4 samples).

It may also be used as a standalone Software Mixer, taking advantage of SSL hardware's near zero latency and high quality DSP processes to create the most flexible, software controlled, hardware console on the planet.

### SSL Soundscape

Soundscape is designed to work as a standalone Digital Audio Workstation, capable of reliably performing recording and sophisticated editing tasks; Its built-in mixer section is identical to the SSL Mixer, but features full automation within the software. In other words, with Soundscape, the Mixer is already built in.

The seamless collaboration between DAW and Mixing functionality inside Soundscape is ideal for mission-critical applications including Live Recording, high channel count Music Production, Broadcast, Post and Restoration.

### SSL Soundscape and SSL Mixer together

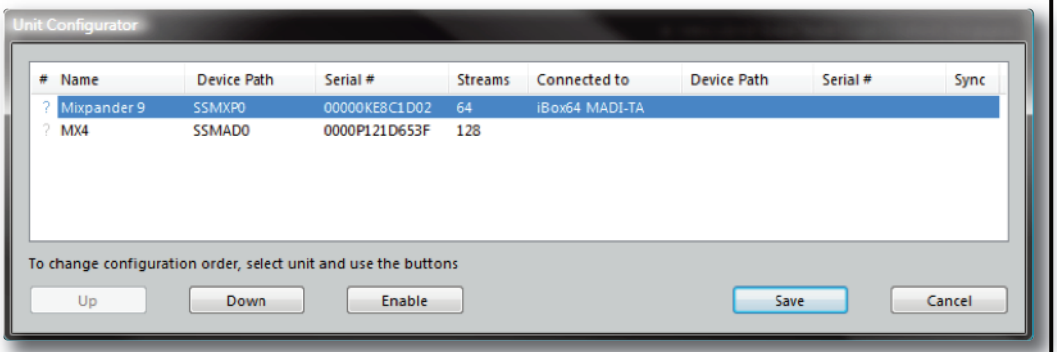
Both software elements may be run simultaneously if necessary – for example, if two different and independent setups are desired. In this way, you might use the SSL Mixer to perform sub-mixing tasks from live inputs or sequencing software, and at the same time run Soundscape to play backing tracks, record live or edit in real time.

*It is important to note that each software entity uses any SSL Card exclusively, so you need at least two SSL cards to run Mixer and Soundscape simultaneously (ie. a MX4 and a Mixpander, both connected to an Alpha Link or MADI iBox interface).*
*To assign each program to work on a different card, you must configure the* ***Unit Configurator*** *utility in the* ***Settings*** *menu of the Mixer and Soundscape, as explained below.*

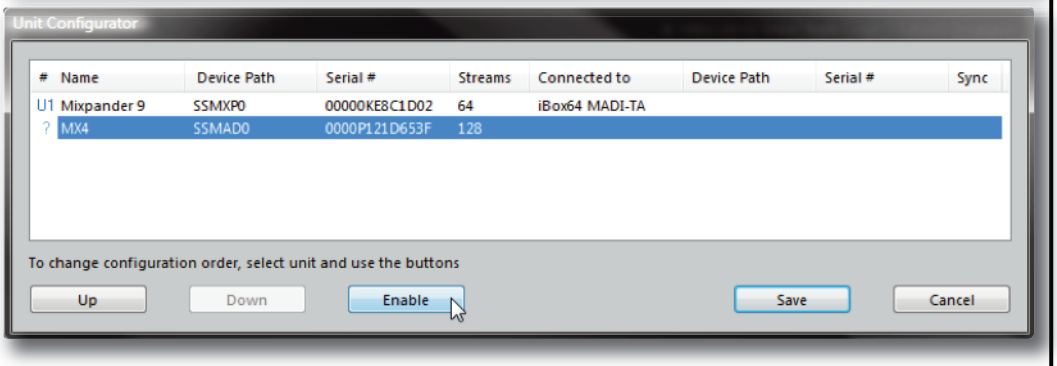
# Unit Configurator

When Soundscape runs for the first time, it will automatically open the **Unit Configurator** (located in the **Settings** menu) to select and configure the hardware you want to use.

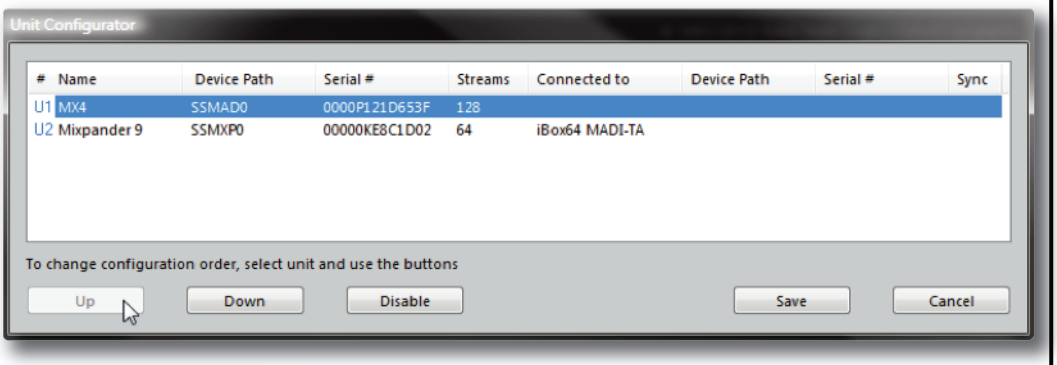
Initially, all SSL units in the system are disabled:




To activate a unit, highlight it and click the **Enable** button:

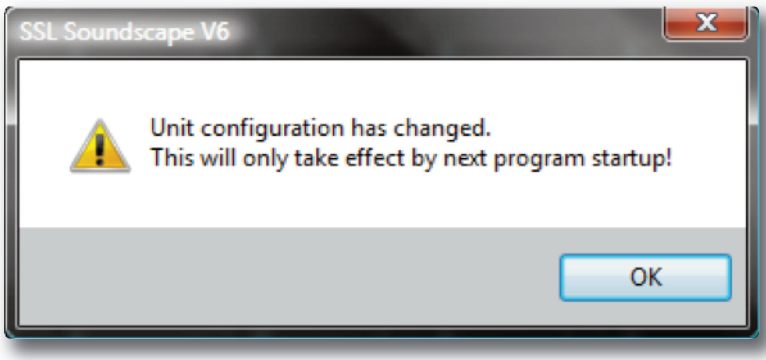



The **Up** and **Down** buttons can be used to renumber units, so that Soundscape lists them in your preferred order. The top Unit is always Unit 1, the second Unit always Unit 2 and so on.




When you are happy with the unit configuration, press **Save**.

A message will appear, warning you that the software must be restarted in order for your configuration to become active.




Press **OK** to close the pop-up, then close and re-open the software. Soundscape will now launch with its Main Window open.

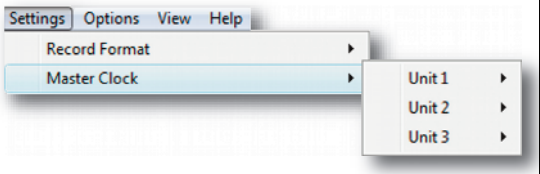
*If your configuration doesn't work as expected, re-launch the **Unit Configurator** from the **Settings** menu.*

# Basic Configuration

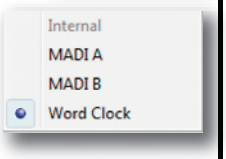
## Clock Synchronisation

Any digital audio system must be properly clocked to a common clock source. In a single Alpha Link configuration, the MX4 or MadiXtreme needs to be locked to the Alpha Link or vice versa. When both the soundcard and the Alpha Link are clocked internally (=no Synchronisation of the clocks) you will hear audible clicks whenever the MADI Frames get too far off.

Select **Master Clock** from the **Settings** menu to configure the clock source for the sound card:

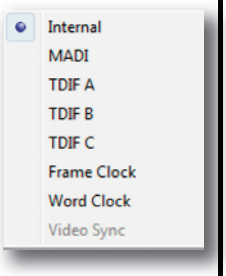



In a system with a single MX4 or MadiXtreme, the choices on the right are available. To make the soundcard the clock master, select **Internal**. Alternatively, to make it a clock slave to the connected audio I/O, choose **MADI A**, **MADI B** or **Word Clock**.




A connected Alpha-Link needs to be clocked appropriately. If the soundcard is set to **Word Clock**, a BNC Cable from the Alpha-Link's Word-Clock Out needs to be connected to the soundcard's Word-clock connector.

To make a **single Mixpander** system the clock master, select **Internal**. Alternatively, set it's clock to be slaved to the connected Alpha-Link by simply choosing one of the digital input ports.




Available inputs will vary depending on the Alpha-Link, iBox or legacy Soundscape 32 connected to the Mixpander's expansion port.

Master Clock can be transmitted and received in a number of ways, depending on the type of cards and converters used in your system. The Word-Clock connectors are an obvious option. MADI, Adat, TDIF, or AES/EBU are also suitable for clocking when two or more SSL Cards are connected to appropriately equipped SSL Alpha-Link Converters.

The Soundscape and Mixer software can work with any combination of MadiXtreme, MX4 and Mixpander cards. Within the Mixer environment, the cards are identified as Unit 1, Unit 2, Unit 3 etc. All cards in a multiple unit system must also be synchronised to a common Master Clock signal which can be provided by one of the cards or by an external device.

*NOTE: In a simple Setup of one MX4 Card plus Alpha-Link Converter, it is advisable to always use the converter's Clock as a Master. Set MX4 to Master Clock **MADI** or **Wordclock**, and the Alpha-Link to **Internal** Clock.*

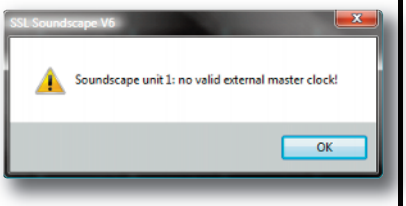
For a multiple Mixpander system, units 2 and above must be operated as a clock slaves, receiving the Clock signal from another unit via their Word Clock or MADI input. The unit that provides the Clock signal must be set to output **Worm Clock**. Worm Clock is a modified Word Clock signal that includes sample-accurate start/stop synchronisation information for the unit via the SSL Alpha-Link Wordclock I/O. The Master Clock parameter for the slave unit must be set to **Word Clock**, which also enables Worm Clock synchronisation.

For more information on Master Clock settings, please refer to the *Master Clock* and *W/S Clock Out* sections of the *Menu Reference* chapter of this manual.

*NOTE: Only use the Worm Clock setting when the clock signal is transmitted to an SSL Alpha Link (or iBox 24/48/64). Other SSL hardware devices may not synchronise correctly if they receive Worm Clock.*

At first start-up, the Master Clock parameter for the master unit (Unit 1) must be set to **Internal** mode or a working external Clock, and the Master Clock for all the slave units should be set according to the current digital connections. This ensures that all units are working from the same Sample Rate clock for sample accurate synchronisation.

If the dialogue box shown right appears, check your digital connections, as the unit indicated is not receiving a valid clock signal at its selected digital input.




Please also refer to the *Master Clock* and *W/S Clock Out* sections of the *Menu Reference* chapter of this manual .

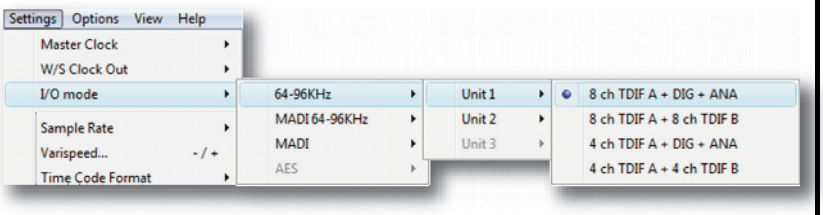
***IMPORTANT:*** *Legacy Soundscape hardware must always be configured as Unit 1.*

## I/O Mode

The **I/O Mode** entry in the **Settings** menu provides several hardware-specific setup submenus for operation with high sample rates and/or high MADI channel counts. The available options are determined by the SSL hardware you are using.

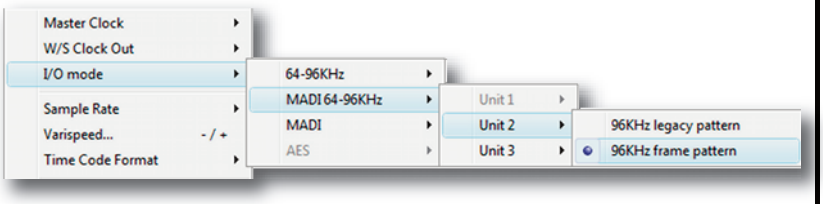
### 64-96kHz

This submenu allows the selection of different I/O configurations and interleaving options at high sample rates, such as SMUXed or non-SMUXed modes for ADAT/TDIF/MADI/AES, or different I/O options for a Soundscape 32/Mixpander system.



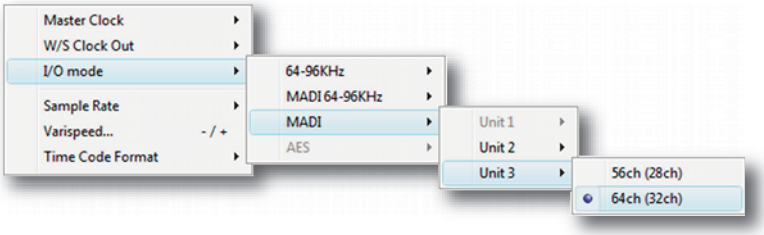

### MADI 64-96kHz

In this submenu, the MADI Frame Pattern is selected between **96k Legacy Pattern** and **96k Frame Pattern**. The options determine how the 28/32 MADI Channels are encoded inside one MADI Frame.




### MADI

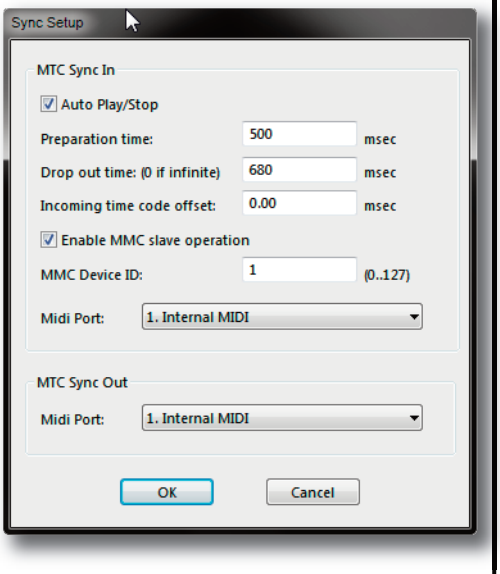
This submenu selects the MADI channel mode between 56 channel and 64 channel.




*Note that by default the SSL Alpha Link's MADI is set to 56Ch MADI mode.*

## Timecode Synchronisation

Timecode synchronisation is managed via the **Sync Setup** display, located in the **Settings** menu. Incoming MTC and MMC, and outgoing MTC can be enabled and configured.




For systems with a Soundscape 32 card, synchronisation information is passed to Units 2 and above by setting each unit's **W/S Clock Out** to **Wormclock**.

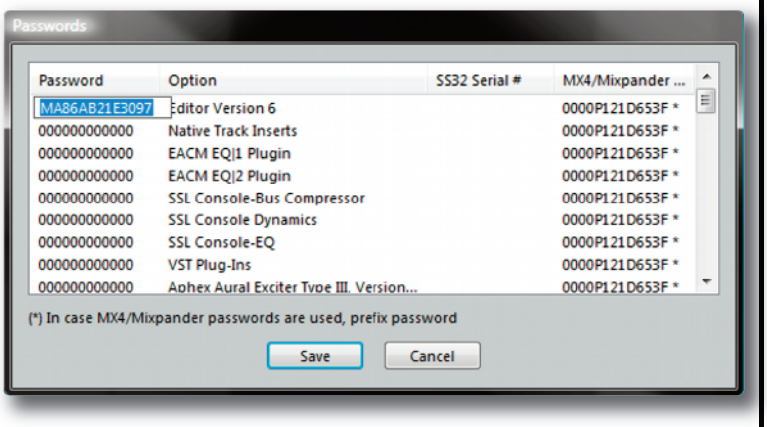
*For detailed information about the **Sync In** options, please read the Synchronisation section in the Menu Reference chapter of this User Guide.*

## Authorising the Soundscape Software and Modules

Click the **Options** menu to open the **Passwords** window. Enter your password for the 'SOUNDSCAPE Version 6', 'Version 2.xx Mixer' (MIX), Audiotoolbox (ATB), along with any other plug-ins such as SSL Console EQ and Filters (SEQ), SSL Console Dynamics (SCD) and SSL Bus Compressor (SBC).

*The Passwords for your SSL Soundscape hardware can be found on a label on the card itself or on the registration card inside the MX4 box.*

If you have registered your units on our website, they are also available in your MySSL account:
store.solidstatelogic.com/user




*If you have purchased any optional plug-ins, this is also the place where you enter their passwords. Please check the 'Options Menu' section of the 'Menu Reference' chapter of this User Guide.*

***IMPORTANT: For MX4 and Mixpander, all passwords need to be entered with the prefix 'M'.***

Entering an incorrect password will result in the following:

- V2. Mixer (MIX): You will not hear any sound or be able to record or play back. Soundscape will notify you with an **Invalid Optional Module Password** message and deactivate the Mixer (Mixer Inactive).
- Soundscape (SS6): Without a Password, the Soundscape Software is limited to Recording and Playback of the first 16 Tracks (TRK 1-16). Everything else is fully functional and you can explore the power of Soundscape. However, if you haven't purchased Soundscape, you will also not have received the included Time Module Password, hence all Timestretch/Pitch Shift/Sample Rate Convert operations are not functional.
- MX4 Included Plug-Ins (ATB, SEQ, SCD, SBC): You will not be able to use any of the these plug-ins. If you load the default or any Standard Mixer or insert one of those Plug-Ins into a channel strip, Soundscape will notify you with an **Invalid Optional Module Password** message and deactivate the Mixer.

*NOTE: Passwords are tied to the specific Hardware UID/Serial Number of the card(s) and can only authorise Modules when the specific hardware is present in the in the current UNIT Configuration. The password-protected modules can then be loaded onto any DSP card as long as this specific UID is present.*

# Opening an Existing Mixer and Arrangement

SSL Soundscape uses separate files for the Mixer and Arrangement, allowing for maximum flexibility. To get an idea of how Soundscape works, you may open pre-made Mixer and Arrangement files. You can download sample Soundscape Projects here:

www.solidstatelogic.com/music/soundscape/downloads.asp

In the **File** menu, click **Open**, then locate and load the **Mixer** file and the **Arrangement** file.

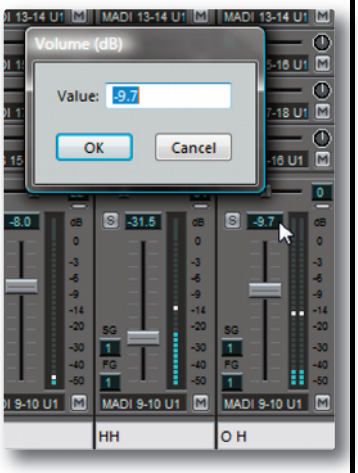
To assign an edit function to a mouse button, click on the appropriate icon located in the Toolbar with the required mouse button. A little black bar will then be displayed under the icon on the corresponding side to remind you which tool is active for which mouse button:



Using the left and right buttons while holding the ***Alt*** key will allow you assign two further tools. These are indicated by a red bar under the icon, on the corresponding side. You will need to press the ***Alt*** key while clicking to use these tools.

When mixing, it might be useful to know that faders and knobs in the Mixer can be controlled three ways:

- by grabbing them with the mouse,
- by positioning the mouse over the controls or value displays and moving the scroll wheel,
- by double clicking on them and entering the desired value.




*NOTE: Several Mixer files are installed with SSL Soundscape. If you installed Soundscape using the default settings, the Mixer file's path is* ***C:\Soundscape\Mix*** *. Selecting* ***Open Mix*** *under the* ***File*** *menu will automatically find this folder.*

*Important: MadiXtreme Cards do not have any DSP's to process Plug-Ins or standard Faders, Sends or Panpots on. MadiXtreme I/O's are also hardwired to the Native Track Element. In other words, Track1 is connected to Input 1 and Madi Output1, and so on.*

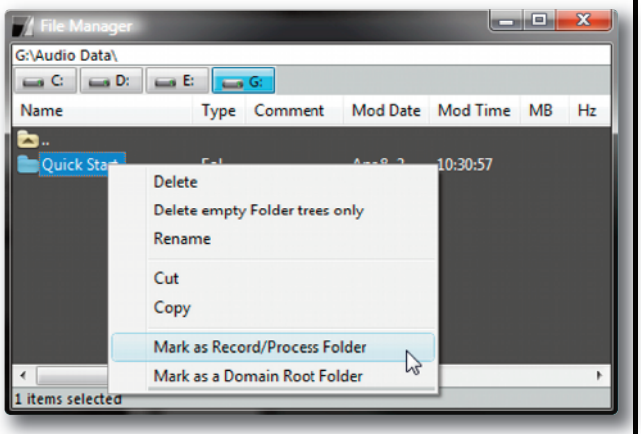
# Creating your first Soundscape Project

## Recording Audio

1. Open Soundscape and navigate to the **File Manager** on the right. (if you don't see the File Manager, press ***D***).

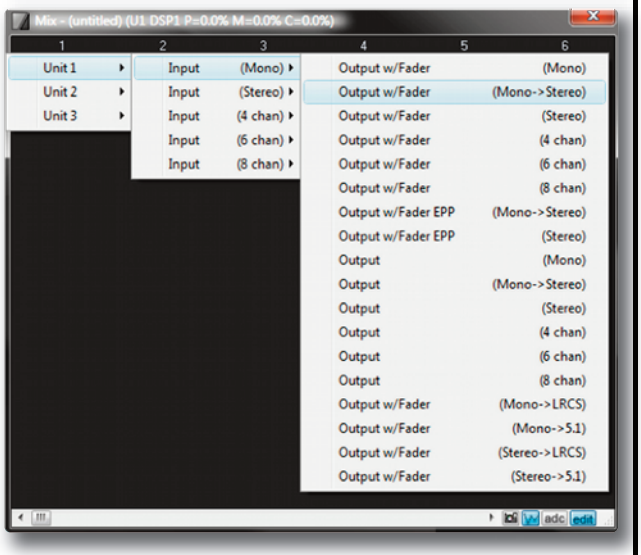
   Navigate to the drive you want to record to, then create a new folder in the **File Manager** by right-clicking in the empty space of the List View and selecting **Create New Folder**.

   Right-click on this Folder (shown as Quick Start in the illustration) and select **Mark as Record** / **Process Folder**, so that Soundscape knows where to store Takes.




2. Create a new Arrangement by selecting **File** / **New Arr...** .

3. Now select **File** / **New Mix...**. You will be presented with a blank Mixer window:




4. Enter **Mix Edit** mode by pressing ***E*** on your keyboard, or by clicking on the **edit** button in the bottom right of the Mixer window.

5. Left-click on the Create tool ( ) and right-click on the Track Assign tool icon ( ).

   This way the two tools needed to create the Mixer are ready for use on your left and right mouse button.

6. Left-click in the first mixer slot of the Mixer window (below slot No.1) and create an **Input (Stereo)** / **Output w/Fader (Stereo)** mixer column.

7. Left-click in the empty area of the new mixer column (below the top Input Label) and insert: **V2.xx Mixer (Part A)** / **Peakmeter (Stereo)**

8. Left-click below the peakmeter to create a native stereo track insert.



**9** Right-click the Input element at the top of the mixer column (assuming the Track Assign tool is assigned to the right mouse button as described above) and select **MADI 1-2** in the pop-up menu (or **Ana 1-2** on Mixpander).



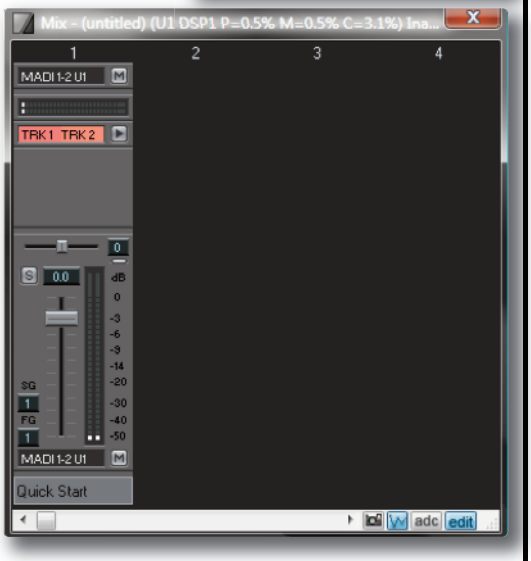

Repeat the procedure for the Output element at the bottom of the column.

Press ***E*** on the computer keyboard to toggle between **Mix Edit** mode and 'Mixer Control' mode (alternatively, click on the Mix Control/Edit toggle, labelled **edit**).

Now double-click in the field at the bottom of the mixer column to open a dialogue box where a name can be entered for the column. For this tutorial we have entered the name 'Quick Start'.

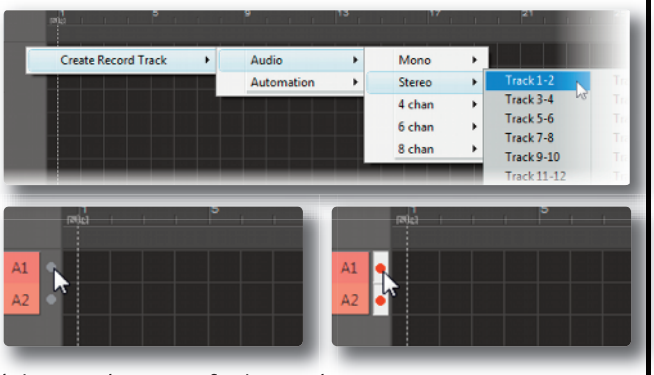
The Mixer should look similar to the one shown right in small view mode (press ***Q*** to show it or hide it):

...and similar to this one in full view mode (press ***X*** to show it or hide it):




**10** Right-click anywhere in the Record Track Column at the left side of the Arrange window.

In the Create Record Track menu which appears, select **Audio** / **Stereo** / **Track 1-2**.




Record tracks 1 and 2 will appear in the Record Track Column with dimmed track arming buttons. Click on the Dots with your left mouse button and they will look 'pushed in' and 'record ready' (because this is a stereo pair, clicking on either dot will activate both record tracks). These buttons are linked to the button on the track insert element of mixer column 1.

**11** Activate the sound source connected to inputs 1/2 of your SSL hardware. The peak meter at the top of the mixer strip in column 1 will indicate the level of the signal at the Input. Make sure the red clipping indicators are not becoming lit. Cue up the audio you want to record and click the Record button in the Tape Transport to start recording (or use the **+** key on your computer's numerical keypad).

*NOTE: If the peak meter indicates no signal or maximum level all the time, the Master Clock settings are probably incorrect. Please check the Master Clock section of the Chapter 5 of this User Guide for details.*

Soundscape will create two temporary Parts (shaded), and will record from the current play position until you press Stop on the Tape Transport (or the ***Down Arrow*** key on your computer keyboard).




When recording starts, the Time Axis will appear shaded in the recording range and the track arming buttons will turn red. The new Takes will automatically be named after the corresponding mixer column, with an 'L' or 'R' appended for stereo indication.

When recording is stopped, the recorded areas, known as Parts, will turn to solid colours and waveforms will be displayed. These Parts can be moved, copied, cut, trimmed, faded, etc., without altering the original Take stored on the Disk.




Let's have a look at the waveform of the audio you recorded:

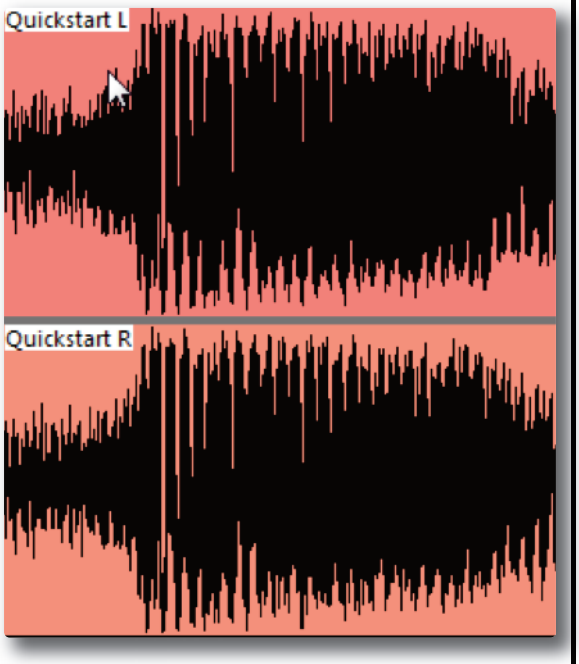
**12** Hit the ***Z*** key on your computer keyboard and the mouse pointer will change into a magnifying glass. If you draw a selection box, Soundscape will zoom to display that selection. You should see a waveform similar to that shown below-right.




*NOTE: Hitting the **Z** key twice will zoom and position to the previous View (Soundscape remembers the last 8 Views). Hitting the **Z** key twice while holding the **Shift** key will zoom and position to the next view. This View History allows extremely fast navigation between detailed sample view and project overview.*

**13** Select the Solo tool from the Toolbar ( ) and click anywhere on the waveform. The audio will play from wherever you click.

Zoom back out to full screen (hit ***Z*** twice to get to the previous view).




**14** Select the Copy tool ( ), then 'click and drag' the Part you have recorded to copy it to a new position in the Arrange window.

Release the mouse button to drop the copy at the chosen location. If the **Snap** function is inactive, you will be able to copy the Part anywhere in the Arrange window. If the **Snap** function is active the copied Part will 'jump' to the nearest snap point, according to the selected snap value.

**15** To name the Part, select the Info tool ( ) and click the Part you want to rename. As soon as the info box appears, you can type in a new name. There are many other details you can alter from within the Part Info window, as described in the *Editing Tools* section of this User Guide.

After typing the new Part name, click **OK** to return to the Arrange window.

## Recording Automation

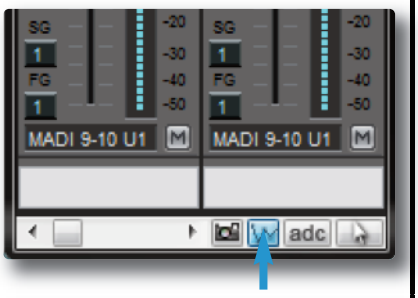
Recording automation data is very similar to audio recording, and shares much of the same procedure. Follow the steps below to see how to record dynamic automation moves.

Recorded automation data can be viewed and edited in pretty much the same way as audio Takes, with the addition of three specific tools which are described in the *Mixer Automation* section in Chapter 4 of this manual. Automation data contains 'automation events', and these come in two varieties: 'continuous events' for faders and pots and 'stepped events' for buttons or 'vintage style' stepped controls.

### 1 Enable Automation:

For any automation data to be recorded or played back, Automation must be enabled. This can be done by clicking the Automation Enable toggle located in the bottom, right corner of the Mixer window, or by pressing ***G*** on the computer keyboard.

When the button is 'illuminated' in blue, Automation is globally enabled.

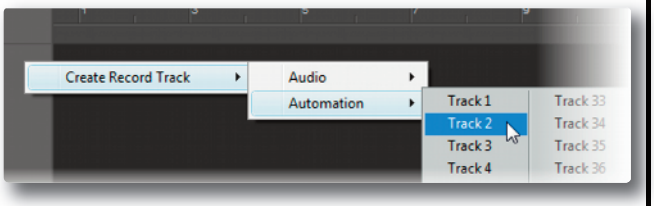



Automation data is recorded on the automation track that matches the Mixer's Column slot number.



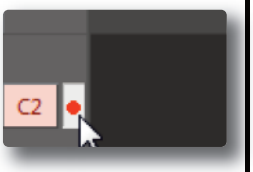

### 2 Create an automation Record Track

In order to record automation, create an Automation record track by right-clicking in the Record Track Column.



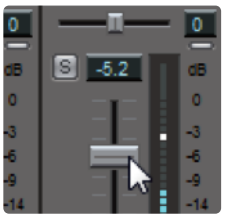

### 3 Arm the Automation Track

The track arming button works just like an Audio Track Arming button.




### 4 Record Automation

Make sure no audio tracks are armed, then start recording automation by pressing the record button or **+** on your keyboard. Now move the desired knobs on the Column Number on which you are currently recording. You can also create multiple automation tracks to record dynamic changes on multiple Mixer Columns.

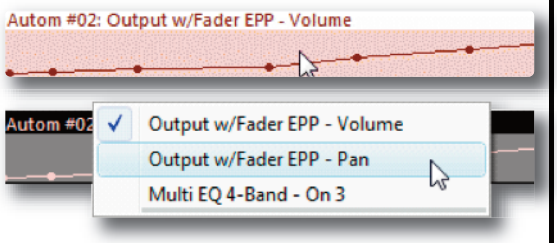



### 5

Press **Stop** to finish the automation recording.

## 6 Curve Select Tool

With the Automation Curve Select tool ( ), you can select which of the automation curves you have recorded is displayed.

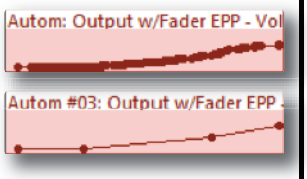



## 7 Automation Events Thinning Tool

After clicking on an automation part, you may enter a thinning factor ( ) to reduce the number of automation nodes/points and make it easier to edit the curves graphically. A thinning factor of 200 reduces the amount of automation nodes slightly, a factor of 1000 reduces the amount of nodes heavily and only maintains the most important nodes (ie. Min or Max of an almost linear Fade In).

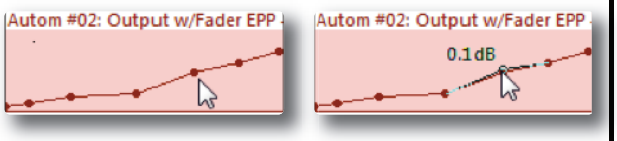
In the example to the right, an automation part was recorded using a motorised fader on an SSL Console to perform an S-Curve like Fade In.

After applying a Thin Factor of 1000, only the shallow slope at the beginning, the steeper slope in the middle and the shallower slope close before the Maximum are maintained in the automation part.




## 8 Automation Event Editing tool

The automation curves may be edited by clicking and dragging the automation nodes with the pencil tool ( ).




*WARNING: Unlike audio Parts, automation Parts can contain several layers of data (automation curves). Simply punching-in on an automation Part (in Normal Record mode) will overwrite all the data in the 'punched' section of this Part (ie., all the automation curves are affected). The* ***Touch Record*** *mode or* ***Touch Record Till Stop*** *mode should be selected for automation punch in/out with no risk of losing any data that is not 'touched'.*

# Migrating Arrangements and Mixes from Soundscape v5 and Earlier

In Soundscape V6 and 7, the proprietary SDisk format, known from earlier Soundscape versions, is no longer recognised. Soundscape now accepts any logical drive recognised by the Windows operating system. This move has beneficits in many areas, such as data transfer and processing speed, faster and easier back-ups, simpler system setups, network collaboration between multiple Soundscape Seats and in general a more user friendly overall experience when dealing with other audio software.

In order to prepare all SDisk Contents to be used with Soundscape V7, the Takes, Arrangements and Mixers need to be copied to standard Windows Drives (FAT16/32/NTFS/NFS etc.).

*Note:*
*For this process, you will need install the Soundscape Editor V5.5. However, you do not need any Soundscape Legacy Hardware. Please follow the instructions in the Soundscape Editor V5.5 Manual on how to setup and connect an SDisk as a Native Unit.*
*If you still have a working Soundscape Editor V2.x to V5.5 setup including a legacy Soundscape System connected, you can also use this and follow the instructions below.*

## Copying files from SDisk(s) to Windows Drives

The first task is to copy the files from the SDisks to a folder or hard drive on any logical Windows drive.

*Please keep in mind that the destination drive must have enough free space. If you want to copy the SDisk's content to a Network Drive or Fileserver (ie. a NAS), you need to Map the Destination Folder as a Network Drive with a Drive Letter.*

1. Open the legacy Soundscape Editor
2. Go to the File menu and select **File** / **Export SFile to PC File** / **All SFiles** / **All SDisks**
3. A dialog box will show the total exported files size and ask you to confirm the creation of the complete folder tree on the PC (recommended).
4. Select the destination drive or folder. After clicking **OK**, a progress bar will be shown for each file
5. Once the transfer is complete, you can close the legacy Soundscape Editor and open SSL Soundscape V7, which can now open all your arrangements and mixer files.

*Note that you will need to point File Manager to the right Drive.*

## Useful Tips for a successful Migration

Please read the following notes carefully to avoid any "Migration blues".

- The Copy operation may take several hours depending on the amount of data being transferred and the speed of your system, especially if the original files are stored on SDisk(s) mounted inside a Soundscape 32 or R.Ed. unit.
- If you can, please mount the SDisk's as NATIVE UNITS via USB/Firewire/eSata as internal PC Disks and use Soundscape Editor V5.5. This process is many times faster and does not require any Soundscape Legacy System to be installed, hence can be done overnight or in the background on any Windows PC/Laptop (running Windows XP or later).
- You can find more information on how to create Native Units in V5.5 in the Soundscape Editor V5.5 User Manual, downloadable here:

  www.solidstatelogic.com/music/soundscape/downloads.asp

- Due to the nature of the proprietary SDisk Filesystem, Arrangements from previous Soundscape Editor Versions are not aware of the concept of File Paths for Audio Takes. Older Versions find Takes by searching and indexing for their Unique Take ID.

  When opening an Arrangement from a previous Soundscape Editor Version, Soundscape V7 will first look for all audio takes with the right Unique Take-ID inside the folder (and subfolders) where this Arrangement is stored.

  If certain audio Takes are stored in completely different folders or even different drives, you will need to tell Soundscape where it is allowed to search for those missing takes.

  With the Invention of Domain/Root folders (you can mark up to 8 Drives or Folders as Domain/Root with a right click) you can create a "virtual Soundscape Environment" including all your working drives and directories containing the copies you made from your Soundscape SDisks.

- Since Soundscape V7 needs to open any Take File residing on a Windows Drive to read the Unique Take ID, search times while opening old Arrangements can exponentially grow if you define multiple complete drives as Domain/Roots.

  Domain/Root assignments are stored globally with |Settings |Save Settings.

- Soundscape V7 stores Arrangements in a new ARR Format. This format includes File-paths for Audio Takes.

  Once you have successfully opened a V5.5 Arrangement that took rather long (due to search times), do not forget to Save the Arrangement (in the new Arrangement Format) to avoid any Search times next time you need to open it.

- If you plan to share files between Soundscape V6-7 and V5.1-5.5 Soundscape Editor systems, you may choose **Save Arr As V5.1 compatible format**.

  Arrangements can only be saved as a V5.1 .Arr, when all active Parts are using Takes in the Soundscape Take File Format (.Atak). As soon as .Wav or .Bwf takes are used, this option is greyed out.

- Mixer files can be shared freely between different versions of the Soundscape Editor and also the Mixer from V2.x and above.

---

*IMPORTANT: In Soundscape V7 the SDisk Menu is gone. In order to assign the Record/Process folders for an arrangement when it's opened for the first time, right click on your desired folder and select* ***Mark as Record/Process Folder*** *from the menu.*
*This Folder assignment is stored inside the Arrangement File.*

---

# Mixing, VST Buffers and ADC

Any digital audio processing takes a small amount of time to be performed. When mixing and applying several processes to an audio track (like EQ, compression, etc.) the cumulative delay ('latency') of one track may differ significantly from the rest, causing phase problems and even audible misalignment.

While all Soundscape-compatible DSP Plug-Ins and Mixer Elements (and almost all Soundscape Legacy DSP Plug-Ins) do not need any additional buffer on top of the central DSP buffer inside an MX4 or Mixpander card (4 Samples round-trip In to Out!), any external processing (either a rack mount FX Device connected to the Converter via audio cables or a native VST/VSTi Plug-In inserted into the Mixer) will add latency.

This delay is most noticeable when inserting VST processes, as the audio has to travel to the computer's CPU, be buffered for processing, and travel back to Soundscape.

The amount of buffering by the CPU is can be defined with the **Native mixer elements sample buffer size** option value in the **Settings** / **Preferences** /**Mixer** menu.

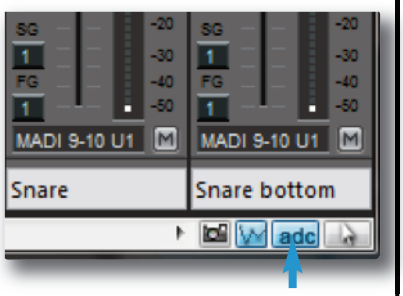
The default value is 128 samples, which should comfortably provide enough time for a modern CPU to process any modern plug-in. If you hear audible clicks or crackles, or feel that the PC is struggling to cope with the processing demands, increase this value in powers of two up to 8192 samples. (ie. 64, 128, 256, 512, 1024...8192)

If an invalid value is entered, it will be replaced with the closest valid value. The Native elements sample buffer size may also need to be increased when mixing very large arrangements with hundreds of VST plug-ins. Since the current VST technology is NOT reporting the loss of samples (=distortion, crackles, clicks) reliably, it is a good idea to set the buffers to the next higher value than appears to be necessary.

However, please note that buffer sizes above 1024 samples can have the reverse effect, since the amount of RAM required to create all the input and output buffers for each plug-in may result in a large system overhead while moving samples between RAM and CPU.

Although the buffer size may be set to a minimum of 64 samples in a modern computer, some sophisticated audio plug-ins may take a longer time to be processed. Also, a chain of VST plug-ins will add an input and output buffer for each one, resulting in multiple processing delays.

Large processing delays can be solved by adding compensating delays to some of the 'faster' signal paths, allowing audio signals to be perfectly aligned at the outputs. To activate the Automatic Delay Compensation on the Soundscape Mixer, click the **adc** button located in the right bottom corner of the Mixer window. When the **adc** button is 'illuminated' in blue, the compensation is active.



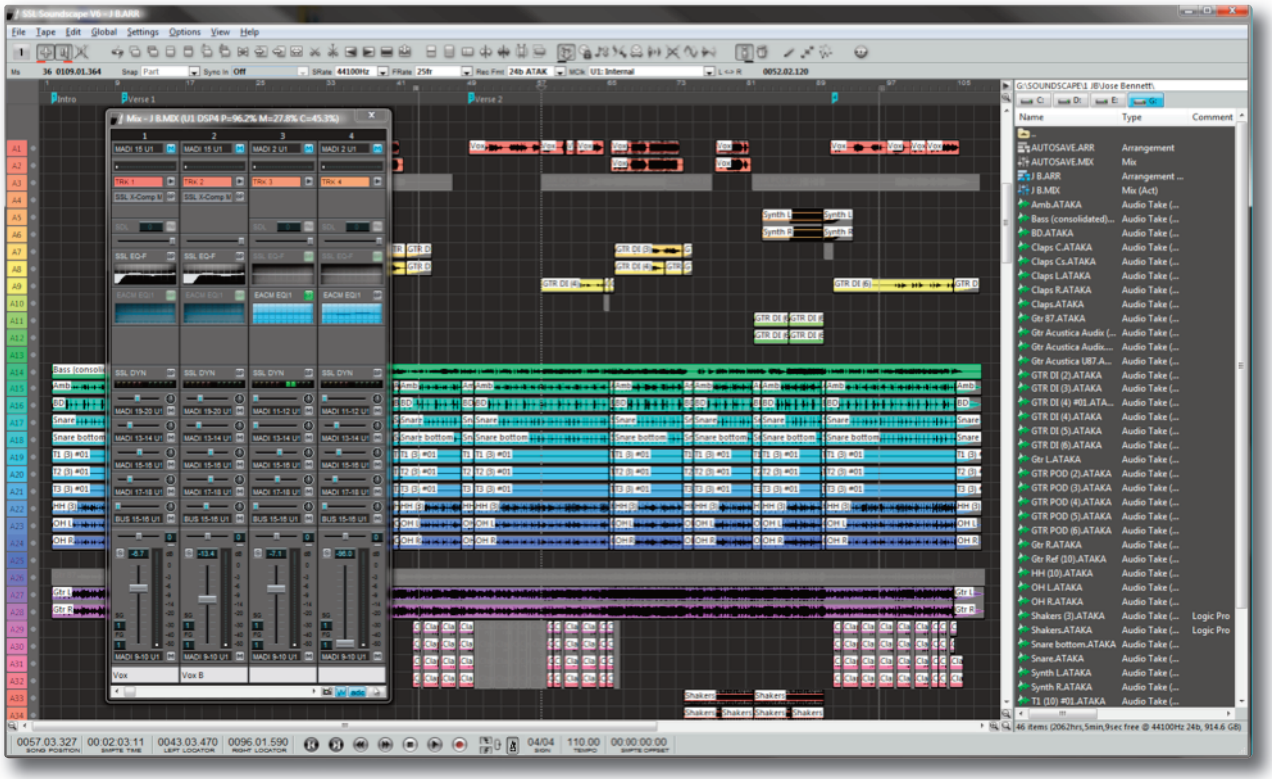

*IMPORTANT: It's not always practical to use Automatic Delay Compensation during recording, since all the signals must be aligned with the 'slowest' one (ie., the one with the highest cumulated processing delay). Any signal going through the system may then be delayed including the artist monitoring paths. Soundscape offers a variety of strategies to exclude certain audio paths (like Monitoring) from being delayed by the ADC. Please read the 'Automatic Delay Compensation' section in Chapter4 of this User Guide for more information.*

# 4. Soundscape Operations

## Main Screen Overview

The main screen is customisable, and you can chose to only display the Soundscape function windows which you find useful. These include the Arrange, Mixer, File Manager, Marker Directory, Video File Player, Big Current Time, Buffer Activity and 9-Pin Sync windows, along with the main Soundscape Window which has the user definable Toolbar and Status Bar across the top and the tape transport bar across the bottom. Each window can be hidden or recalled with a saved size and position, with a single keyboard shortcut or menu selection.

### Docking Arrange and File Manager Windows to the Main Window.



For multi-monitor setups, all Soundscape windows can be freely placed anywhere inside the extended desktop. A Soundscape Timeline Project is called an Arrangement, and has 1024 virtual Tracks to create adventurous projects. It can also contain up to 131072 individual Parts (Blocks of Audio on the timeline) which use up to 131072 audio Takes (Audio Files on the Disk). An Arrangement can be created from scratch, or a saved Arrange file (.ARR) can be loaded.

Soundscape's extremely flexible Mixer can have up to 128 Mixer Columns (or Channel Strips) that can each be Mono, Stereo or up to 8 Channels wide. Mixers are stored separately as .MIX files, and while you are still wondering why the Mix is not stored in the Arrangement, we can assure you that this will actually revolutionise the way you work with different Mix Versions and full A/B comparison complete Mixers.

The Arrange file, Mix file and any associated Audio Takes can be backed up to any logical PC drive, to secure your data.

# TOOLBAR




At top of the main Soundscape window you will find the main menus, the Toolbar – which is used for most on-screen editing – and the Status Bar.

*Note that the main menus are explained in detail in Chapter 5.*

There are nine pages of tools, any of which can be displayed in the Toolbar. Scrolling through the Tool Pages can be done by clicking the page number icon, or by pressing the ***Page Up*** and ***Page Down*** keys on the computer keyboard. The Toolbar page number is shown to the left of the icons (shown right).




*NOTE: It is possible to go directly from Page 9 to Page 1 and back, without scrolling through all the other pages.*

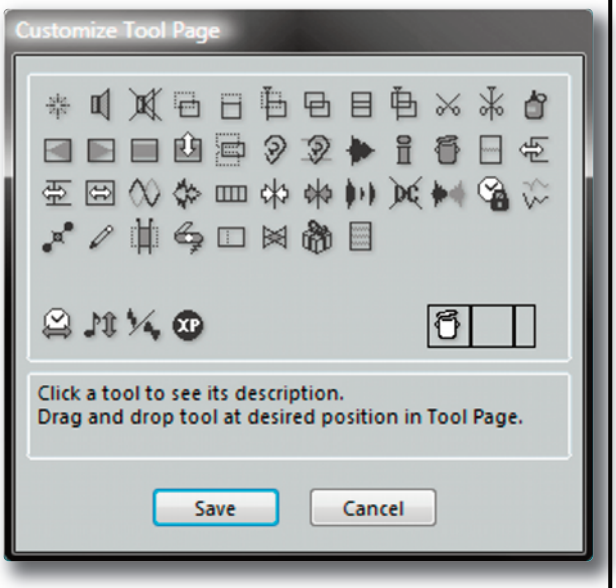
Double-clicking on an empty space in the Toolbar, or clicking **Customize Tool Page** in the **Settings** menu, will open the **Customize Tool Page** window, where the current Tool Page can be customized.

Clicking on any tool icon will display a description of its function. Icons can be dragged and dropped to any position in the Toolbar, and any existing tools will move to make room for it.

To delete a tool from a Tool Page, drag and drop the white trash can icon over it. The other tools will then close up the space.

To insert a space between tools, drag and drop the full space or half space icons (the empty boxes beside the trash can icon).

Once you are happy with the Tool Page, click **Save** to save all changes you have made to the current Tool Page globally.




## Tool selection

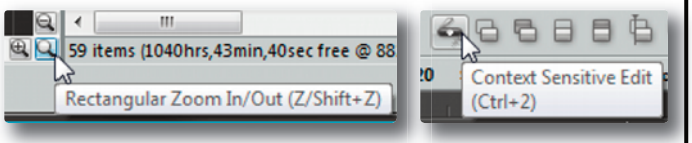



Four tools can be assigned to the mouse buttons, as shown above. Click on the tool of your choice with the mouse button you want to assign it to. A black bar will be displayed below the selected tool's icon, to the left or right side according to the chosen mouse button. You can select two other tools by holding down the ***Alt*** key while you click on the relevant tool icons. These tools are used by holding down the ***Alt*** key and pressing the corresponding mouse button.

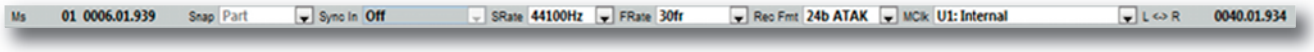
*NOTES:*
*A tool will remain selected even if a different Tool Page is displayed.*
*If a tool is assigned to a mouse button both with and without the **Alt** modifier, only the black bar will be shown*
*Selecting the 'Context Sensitive Edit' tool will cancel the previous selection, as it uses all four mouse functions.*

## Tooltips

When the mouse hovers over an editing tool or other function button, a tool tip is displayed after about a second. Where applicable, its key command is shown.



# Status Bar




The Status Bar, displayed underneath the Toolbar, displays some of the more commonly accessed parameters from the **Settings** menu, and allows them to be selected quickly.

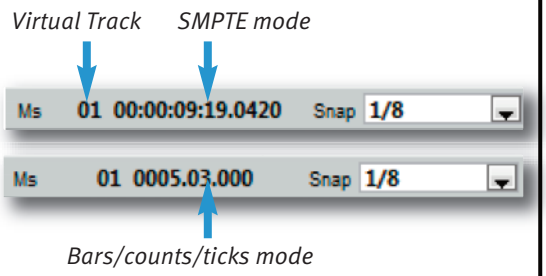
The bar shows the current mouse position within the Arrangement, the current status and selected value of the Snap setting, the Synchronisation mode, Project Sample Rate, Frame Rate, Sample Resolution and Record File Format for new recorded/processed Takes.

The time interval between the Left and Right Locators is displayed in bars/counts/ticks or SMPTE + sample extension at the current Frame Rate.

*NOTE: The Status Bar can be turned on or off in the **View** menu.*

## Mouse position

The mouse position readout is continuously updated to the current position of the mouse pointer in the Arrange window. Its resolution is determined by the **Snap** setting (if active), while the **Time Axis** setting (**SMPTE** or **bars/counts/ticks**) determines the time position's units. In **SMPTE** mode, the sample within each frame is also shown.




The number displayed to the left of the position display indicates which virtual track (horizontal position) the mouse is currently over. This is useful for keeping Parts on the same virtual track in large Arrangements.

The mouse resolution across the display depends on the horizontal screen resolution, so do not expect to be able to move in steps of one frame when you have five minutes of the Arrangement are displayed screen. However if you zoom in, the mouse position readout will be as expected.

The theoretical resolution of musical time divisions is 960 ticks per quarter note, and the actual time value of one tick is variable depending on the Tempo. The resolution of SMPTE time divisions is a 100th of a frame, irrespective of Tempo, but is variable according to the Frame Rate selected under the **Settings** / **Time Code Format** menu.

However, sample accuracy is often required when editing digital audio. If **SMPTE time** is selected from the **Settings** / **Time Axis** menu, the mouse position readout employs a format that makes sample accurate editing with the mouse easy. It is necessary to use the highest zoom in level so that each pixel on the screen shows one single sample, and the **Snap** function needs to be inactive.

The complete format is: **hours:minutes:seconds:frames:samples**, preceded with the number of the virtual track where the mouse pointer is positioned.

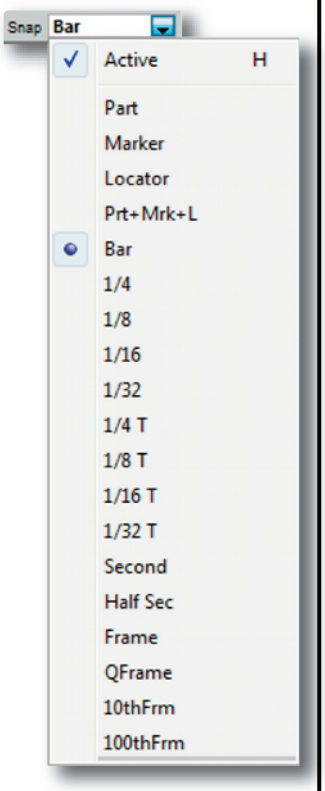
## Samples per frame:

The actual number of samples within each frame depends on the Sample Rate and SMPTE Frame Rate selected. The table below shows the number of samples in each frame for all standard Sample Rate and SMPTE Frame Rate settings. Where the value is not a full number, it will be averaged over a number of frames, so that there will always be a full number of actual samples within each frame.

*Note that 29.97 frame rates listed below are in fact the more accurate 30x1000/1001 calculations*

| Number of samples per frame | | | | | | |
|---|---|---|---|---|---|---|
| | **24 fps** | **25 fps** | **29.97 fps*** | **29.97 df*** | **30 fps** | **30 df** |
| **22.050 kHz** | 918.75 | 882 | 735.74 | 735.74 | 735 | 735 |
| **32 kHz** | 1333.33 | 1280 | 1067.74 | 1067.74 | 1066.66 | 1066.66 |
| **44.056 kHz** | 1835.66 | 1762.24 | 1470 | 1470 | 1468.53 | 1468.53 |
| **44.1 kHz** | 1837.5 | 1764 | 1471.47 | 1471.47 | 1470 | 1470 |
| **47.952 kHz** | 1998 | 1918.08 | 1600 | 1600 | 1598.4 | 1598.4 |
| **48 kHz** | 2000 | 1920 | 1601.6 | 1601.6 | 1600 | 1600 |
| **64 kHz** | 2666.66 | 2560 | 2135.47 | 2135.47 | 2133.33 | 2133.33 |
| **88.112 kHz** | 3671.33 | 3524.48 | 2940 | 2940 | 2937.06 | 2937.06 |
| **88.2 kHz** | 3675 | 3528 | 2942.94 | 2942.94 | 2940 | 2940 |
| **95.904 kHz** | 3996 | 3836.16 | 3200 | 3200 | 3196.8 | 3196.8 |
| **96 kHz** | 4000 | 38.40 | 3203.2 | 3203.2 | 3200 | 3200 |

## Snap




The **Snap** function ***H*** defines a global time grid for almost any editing and positioning function. Clicking on the **Snap** drop-down box opens the list of possible values.

With **Snap** active, time position markers as well as Left, Right and Current Locators, will automatically snap to the next point in the grid, and edits will be performed on that Snap Point. This allows fast and accurate editing without the need to zoom. For example, when **Snap** is set to **Bar**, cuts will be made at the barline nearest the mouse position.

The **Snap** function is also useful for positioning the Locators for punch in/out recording exactly on beats. For example, if you want to edit the audio on quarter-bars, set **Snap** to **1/4**.

While working with picture, setting **Snap** to **Frame** can be used to quickly place audio clips exactly as required.

If a part is moved, its start will be placed at a Snap Point. You can drag the new Part with the **Copy** tool, and as long as you release the mouse close enough to the target position it will snap into position perfectly.

The first item, labelled **Active**, allows activation/deactivation of the Snap function. The ***H*** key on the keyboard toggles this setting.

The currently selected Snap value appears black in the drop-down box if the **Snap** function is active; It is dimmed if inactive.

*Note that with Snap active, the mouse position readout shows the time code of a potential edit, not the precise mouse position. (please see above)*

Certain editing tools – such as the **Move Vertical** or **Copy to Locator** tools – are not subject to the **Snap** function. The **Slip** and **Repeat** tools respond to the **Snap** function only if the selected Snap value is a musical or SMPTE based time division, as described later.

If the selected Snap value is **Part**, **Marker**, **Locator**, or **Prt+Mrk+L**, the beginning and end time positions of all **Parts**, the **Markers**, the Left, Right, and Current **Locators**, or all of these together (**Prt+Mrk+L**) will respectively act as snap points whenever a snap-sensitive edit or operation is performed close enough to one of them, along with any embedded Snap Points (as described in the *Snap Point Edit tool* of the *Editing Tools* section in this chapter). If the position for an edit is out of range, the Snap function will have no effect.

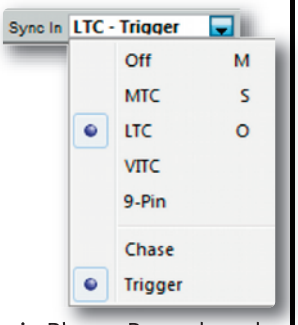
In the above cases, the same 'object' could be subject to the Snap function and also act as a snap point itself. For example, if the selected value is **Prt+Mrk+L**, a Part's beginning could be snapped to the Current Locator, and then the Current Locator could be snapped to the end of that Part.

If the Snap function is active and the selected snap value is a musical time division (**Bar**, **1/4**, **1/8**, **1/16**, **1/32**, **1/4T**, **1/8T**, **1/16T**, **1/32T**), or a SMPTE or SMPTE-related time division (**Second**, **Half Sec**, **Frame**, **QFrame**, **10thFrm**, **100thFrm**), then all snap-sensitive edits and the positioning of the Markers, Left and Right Locators, or Current Locator always occur at the closest snap point. There is no 'safe distance' from the snap points.

*NOTES:*
*The Markers, Left/Right Locators and the Current Locator can be set to any time position regardless of the snap settings when entering time code values with the keyboard. 'On the fly' Locator dropping using key commands to place Markers or Locators, is not snap-sensitive either.*
*If the active snap setting is **Part**, **Marker**, **Locator**, or **Prt+Mrk+L**, both beginning and end of the edited Part will snap.*
*Please also read about the Snap Point Edit tool in the Editing Tools section of this chapter.*

## Sync In

The **Sync In** box displays a list of 'Synchronisation Slave' modes also found in the **Settings** / **Sync In** menu. Options are: **Off**, **MTC**, **LTC**, **VITC**, and **9-Pin**.




Pressing ***M*** on the computer keyboard sets the **Sync In** mode to **Off**, allowing Soundscape to be used as a Time Code Master.

**MTC** and **LTC** can be selected by pressing the ***S*** and ***O*** keys respectively. **LTC**, **VITC**, **BITC** and **9-Pin** are only available if the optional Soundscape Sync Board is installed in a legacy Soundscape 32 unit and this Unit is present as Unit 1.

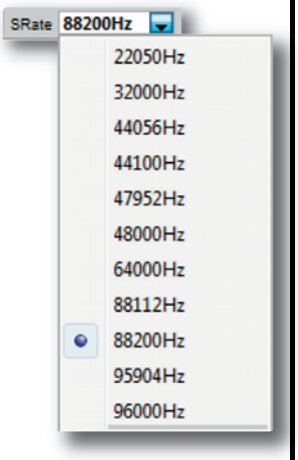
The **9-Pin** option is only available if **Controller**, **Synchro** or **Layback** is selected in the **Settings** / **9-Pin Operation Mode** menu.

In **Chase** Slave or **Trigger** Slave modes, Soundscape will only respond to incoming Time Code if it is in Play or Record mode (ie., if the **Play** or **Record** button has been clicked) unless the **Auto Play/Stop when slave syncing** option is enabled in the **Settings** / **Sync Setup** window.

> ***IMPORTANT:*** *External Time Code synchronisation is only available if there's a Soundscape 32 unit in the system.*

## Sample Rate

Clicking the **SRate** Drop Down box will display all available sample rates. Clicking on an entry will set Soundscape to operate at that rate. This can also be specified in the **Settings** / **Sample Rate** menu.

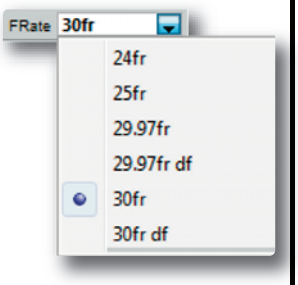



When the Master Clock is set to an external input in the **Settings** / **Master Clock** menu, the **SRate** readout displays the sample rate being received. The display will appear dimmed, indicating that the sample rate selection cannot be altered. Soundscape operates at sample rates up to 96kHz.

Whenever the **Varispeed** value in the **Settings** menu is not equal to 00.00%, the sample rate will be displayed in red.

## Frame Rate

Clicking the arrow to the right of the **FRate** box will open a list with available frame rates.

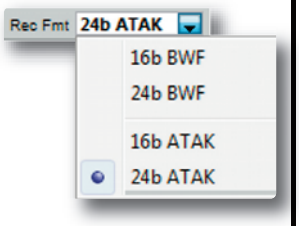



This parameter can also be set in the **Settings** / **Time Code Format** menu.

> *Note that 29.97 frame rates are in fact the more accurate 30x1000/1001 calculations*

## Record Format

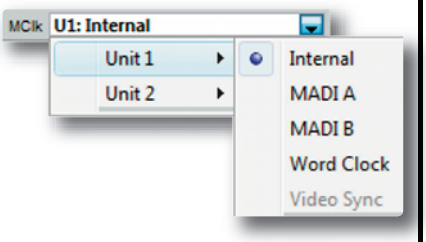
The **Rec Fmt** box shows Soundscape's current recording file format and bit resolution. Clicking on its drop-down box allows you to select between 16 or 24-bit BWF broadcast wave file and 16 or 24-bit ATAK SSL Soundscape audio take format. Changing the format will only affect newly recorded or processed audio Takes.




This option is also available in the **Settings** menu.

## Master Clock

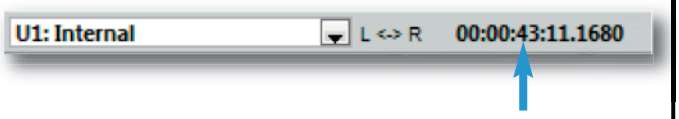
The **MClk** box shows the current master clock settings. The clocking options are shown for each unit present in the system.




This option is also available in the **Settings** menu.

## L<–>R Readout

The **L<–>R** readout shows the current time interval between the Left and Right Locators.

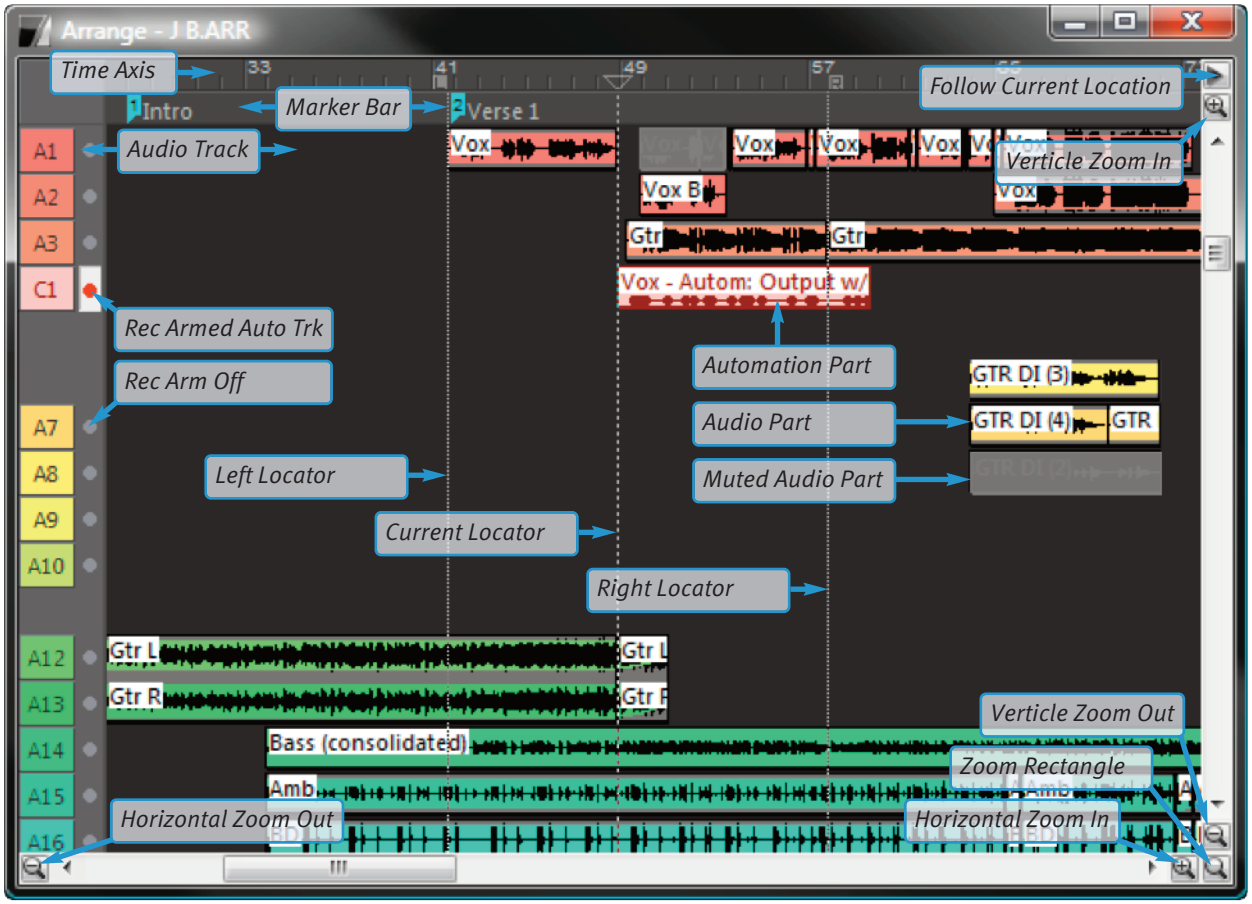



The value is displayed in bars, counts, ticks or SMPTE+sample extension, according to the setting chosen for **Time Axis** in the **Settings** menu.

# Arrange Window

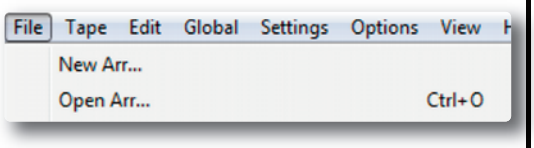
The Arrange window is where most of the real work happens. It shows the layout of your Arrangement, with up to 1024 virtual tracks displayed vertically. Active audio Parts are colour-coded when assigned to one of the 128 physical tracks (64 with one Mixpander). Active automation Parts can use any of the 128 automation tracks which are connected to the Mixer Column Number.

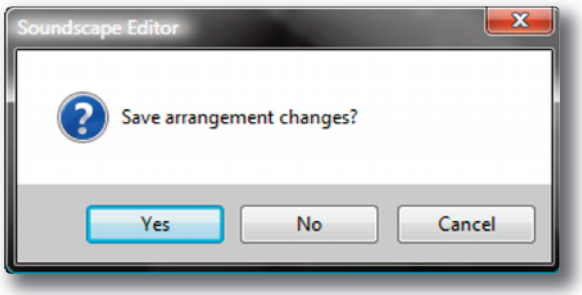
Muted Parts appear greyed out with, their name and/or audio waveform/automation events dimmed; They are currently not assigned to an audio or automation track. When the Arrangement is played back, therefore, muted audio Parts will not be heard and automation Parts will not control Mixer Elements.




## Creating and Opening Arrangements

To create a new, empty Arrangement window, select the **File** menu's **New Arr...** option.

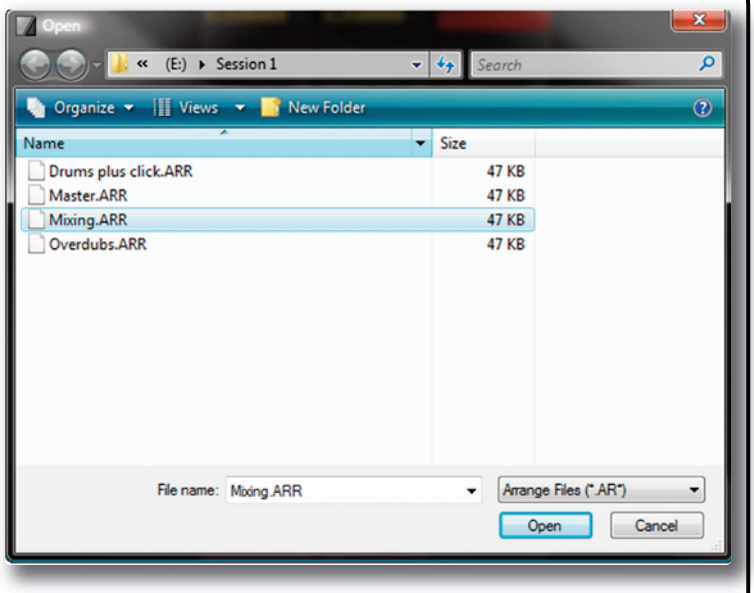



If an Arrangement is already open and has unsaved changes, you need to save or lose those changes, or **Cancel** the operation and revert to the active arrangement.

Selecting **Open Arr...** (***Ctrl+O***) brings up a standard Windows file browser.

Again, if an Arrangement is already open and has unsaved changes, you need to save or lose the changes, or **Cancel** the operation completely to revert to the active arrangement.




*Arrangements can also be opened by double clicking an ARR file inside File Manager.*

## Appending Arrangements

**Append Arr...** (***Ctrl+A***) adds a saved .arr File to the currently opened Arrangement at the Current Locator position.

*Note that the Current Locator should be placed at the desired song position before using this option.*

If Part overlaps are not allowed (As set in **Settings** / **Preferences** / **Arrangement** / **Overlapping Parts (for new edits)**), any Parts in the appended Arrangement which would overlap with Parts in the current Arrangement will not be loaded.

Arrangements can also be appended and visually positioned by dragging and dropping an Arrangement from File Manager.

## Saving Arrangements

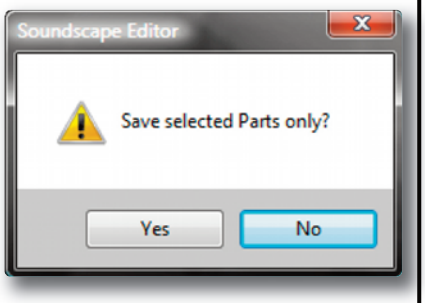
**Save Arr...** (***Ctrl+S***) saves the current Arrangement. Arrangements don't only store the list of Parts on the timeline, but also include project settings such as sample rate, frame rate, tempo, SMPTE offset, varispeed, L and R locators, markers, punch and loop button status, record setup, automation setup, and the assigned Record/Process folder.

**Save Arr As...** allows the Arrangement to be saved under a different name, using a standard Windows dialogue.

In order to keep compatibility with legacy Soundscape Systems, Arrangements can be saved as a Soundscape Editor V5.1 compatible format using the **Save Arr As V5.1 compatible format...** option. This is only available if no WAV/BWF files are used by Parts inside the Arrangement.

### Saving selected Parts as an Arrangement

If any Part is selected in the Arrange Window when you click **Save Arr As...**, a dialogue box will appear (shown right).




This function allows you to only partially save an Arrangement and therefore create Multi Channel pieces which can be used in other Arrangements (using **Append Arr...**).

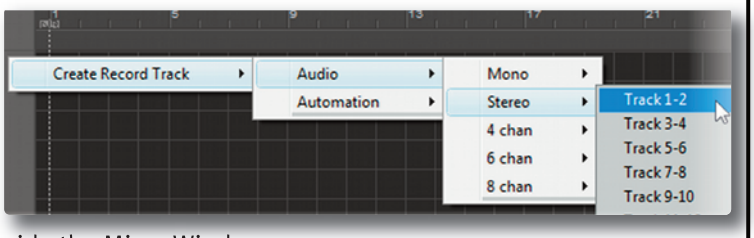
This can be useful for recurring Jingles, Openers or Sound FX.

Click **No** if you want to save the whole Arrangement.

## Record Tracks

The Record Track Column allows you to determine the virtual track position, record loop stack size and track arming status for the physical tracks. Any record track inserted in the Record Track Column can also determine the default output track assignment for the corresponding virtual track when dragging Takes into the Arrangement or copying/moving Parts to a different virtual track.

Since Soundscape does NOT have a fixed relationship between virtual track and output assignments, record tracks can be freely moved without changing the track assignments of Parts that are already on a virtual track.




Right-clicking on the Record Track Column opens a menu where you can create an audio or automation record track on this virtual track (there can be up to 256 virtual tracks). Audio record tracks can be created as mono tracks, stereo pairs or multichannel groups.

For automation record tracks, please note that you select the Column Slot Number you want to automate inside the Mixer Window .

When using an MX4, there is also no fixed relation between track I/O assignments and the Mixer Column Number in Soundscape, hence Track 1 Audio (A1) could be connected to Mixer Column 10, making it necessary to create automation for this audio track on Automation Track C10.

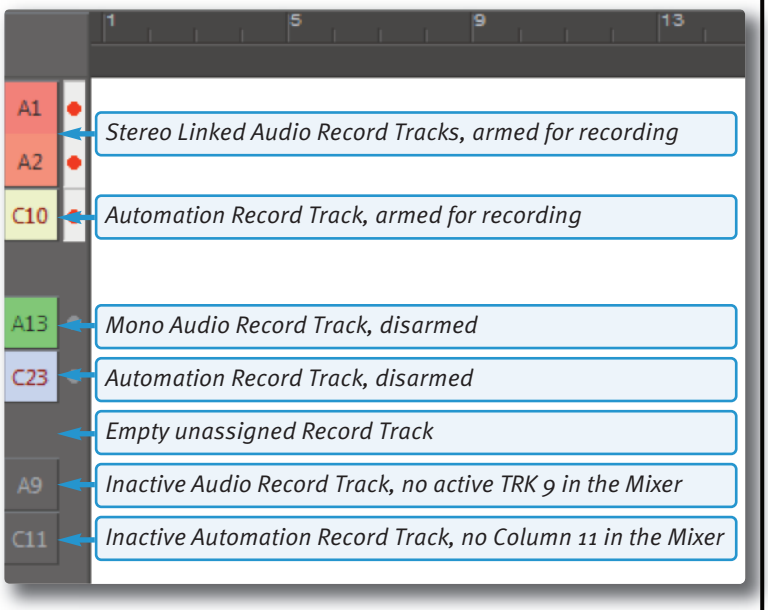
*Note that with a MadiXtreme, there is a fixed relationship between I/O assignments and Mixer tracks.*

Baffled? Confused? The good news is, you can assign Audio Track 1 to virtual Track 1 and Mixer Column 1... if you like to keep it simple!

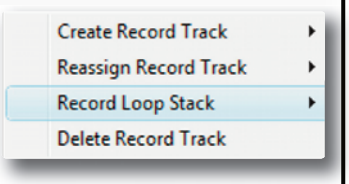
Once created, record tracks appear as colour-coded slots in the Record Track Column, with a round 'track arming' button.

If an audio record track has no corresponding track insert in the Mixer, or if an automation record track has no corresponding mixer column, the rectangle will be dimmed and the track arming button will be absent.

*NOTE: A mono Audio Record Track cannot be used if it is connected to a Stereo Track Insert in the Mixer. A stereo Audio Record Track, however, will enable both Mono Track Inserts in the Mixer.*

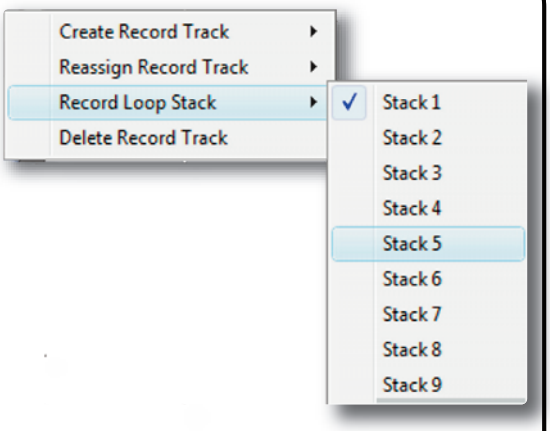



Right-clicking in a slot with an existing record track opens a slightly different menu. You can still create an audio or automation record track (the new track then just replaces the existing one), or you can reassign an existing record track (ie., change its Track number), create a record loop stack (for loop recording) or delete an existing record track.




If you want to replace an automation track with an audio record track, simply select **Create Record Track** / **Audio** / **Trk...** in the menu. If you select **Reassign Record Track**, you can change a track's number but not its type.

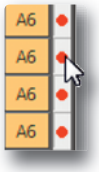
If you right-click on an existing record track which is already part of a loop stack, selecting **Create Record Track** will only replace the existing track in the slot you clicked on, without affecting the rest of the loop stack. Selecting **Reassign Record Track**, however, will change the track number for the whole stack.




Any record track or record loop stack can be moved to a new position in the Record Track Column by dragging it with the left mouse button.

Several record tracks can be selected at the same time by holding down ***Ctrl*** and clicking them one by one, or by holding down ***Shift*** and clicking the first and last tracks in a range. Multiple selected tracks can be dragged as a group to new virtual track positions, or deleted.

If you drop a record track or record loop stack on an existing record track, the existing record track disappears.




The Undo function can be used to reverse these changes.

A record loop stack can be armed for recording by pressing any of the track arming buttons within that stack.

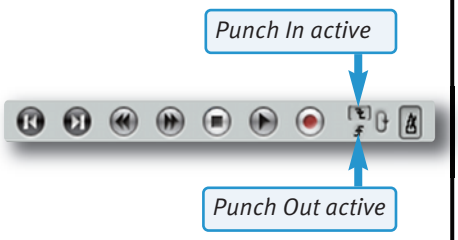
## Recording Audio

1. Soundscape opens **Soundscape def.mix** by default. If your Mix Window is empty, you need to open a Mixer for recording (**File** / **Open Mix** or double click on a **.mix** file inside the File Manager Window) or create one from scratch.

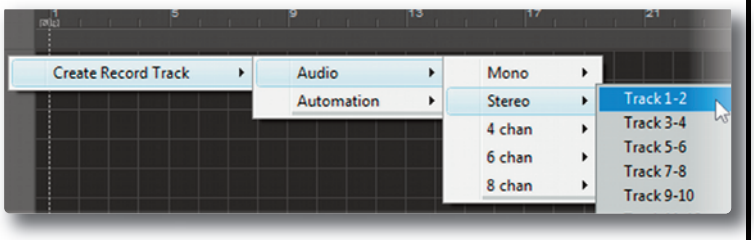
2. If you want to use the Auto Punch In/Out, place the Left and Right Locators at the desired Auto Punch In and Punch Out time positions by clicking inside the Time Axis with the left mouse button for the Left Locator and the right mouse button for the Right Locator. You can also move the Left and Right Locators by clicking and dragging them with the corresponding mouse button in the Time Axis.

   Enable the Auto Punch In and/or Auto Punch Out function(s) using the buttons in the Tape Transport Bar. (see right)

   If Auto Punch In is inactive, the Current Locator determines the recording start location (Crash recording). The Current Locator can be repositioned by clicking at the time position in empty space in the Arrange window, or by ***Shift+*** clicking inside the Time Axis.



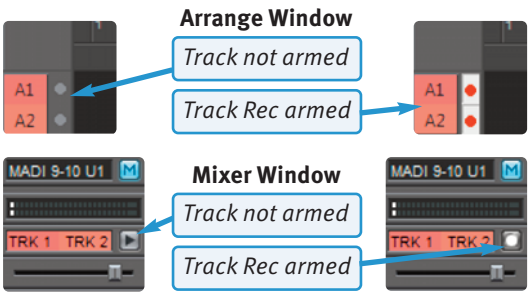

3. Right-click on a free slot in the Record Track Column. A menu will appear that lets you select mono or stereo and which tracks to record onto. Release the mouse button on the number of the track(s) you wish to record onto. If you are using **Soundscape def.mix**, select **Stereo** and choose **Track 1-2**.




4. Arm the track(s) by clicking the track arming button(s) to the right of the track numbers.

   A record track can only be armed if there is a corresponding track insert element in a mixer column.

   Existing record tracks can also be armed from the Mixer, as the track arming button is duplicated in the corresponding Track Insert element. Record Tracks and Track Inserts share a colour for quick and easy visual identification.



### Audio record track arming using the computer keyboard's numerical keypad

You can also arm or disarm individual or multiple audio record tracks directly from the keyboard by using the number keys, along with ***** to arm, **/** to disarm and **–** to select a track range.

*NOTE: Only record tracks that have already been created in the Record Track Column and have a corresponding track insert in the Mixer can be armed.*

- The ***** key arms all the available audio record tracks
- A number key followed by ***** arms the corresponding audio record track (eg. ***5**** arms track 5).
- Pressing two number keys separated by the **–** key and followed by the ***** key arms all audio record tracks from the first to the second number (eg. ***3-8**** arms all existing audio record tracks from track 3 to track 8).
- The **/** key disarms all previously armed audio record tracks.
- A number key followed by the **/** key only disarms the corresponding record track.
- Pressing two number keys separated by the **–** key and followed by the **/** key disarms all audio record tracks from the first to the second number (eg. ***3-8/*** disarms audio record tracks track 3 to 8).

*NOTE: It is not possible to arm a mono audio record track if it is designated as part of a stereo pair in the active Mixer. For example, if tracks 1 and 2 are a stereo track insert mixer element, it is not possible to only record on track 1 or only on track 2.*

Regardless of the audio track used for recording, it is very easy to assign recorded audio Parts to different track numbers for playback. Therefore, the creation of a specific record track for any Playback Track may not be very important.

5 Repeat the record track creation and arming procedures for any other tracks you wish to record to, and check the input levels (if you are using Mix3.mix, use the peakmeters at the top of the mixer columns).

*WARNING: Be very careful not to cause the input to clip while recording, as with all digital systems, this can result in very high distortion levels, especially for full-scale Input longer than half a dozen of samples. Digital Systems do not have a concept of 'Head-Room'; It is therefore a good idea to leave some artificial headroom while adjusting your Input levels and probably using some light limiting and compression when recording very dynamic sources.*

6 Click the Record button on the Tape Transport or press the **+** key on the keyboard to start recording.

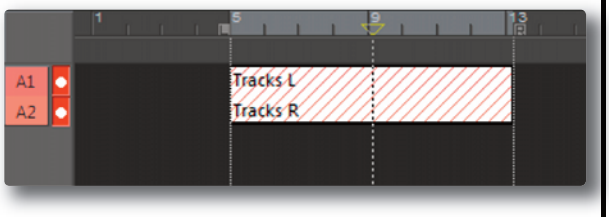
*NOTE: The Record button can be used to toggle between Play and Record modes, allowing multiple consecutive manual punch in/outs to be performed in one single pass (but note that if **Auto Punch In** is active, it also causes the Current Locator to jump back to the Left Locator (minus preroll) each time the Record mode is activated).*
*Clicking the Play button toggles from Record mode to Play mode. If the **Allow track arm changes during record w/gapless punch-out** box in the **Settings** / **Record Setup** window is checked, the track arming buttons can also be used in Record mode to perform manual punch ins and punch outs for individual tracks (or groups of tracks).*
*Please read the 'Record Setup' section of the 'Settings Menu' in Chapter 5 for more details.*

If the range you have designated for recording overlaps a Part with the same output assignment on another virtual track, the overlapped section of the active Part will be muted. If the **Overlapping Parts (for new edits)** option under **Settings** / **Preferences** / **Arrangement** is set to **Not allowed**, existing Parts underneath the new recorded Part (ie., on the same virtual track) will be cut and deleted. If it is set to **Allowed**, newly recorded Parts are stacked on top of the existing ones.

The default recording time for manual recordings can be changed in **Record Setup** (**Settings** menu) and via the **Record Setup** window. By default, this will be set to 0 minutes, allowing unlimited recording.

7 If **Auto Punch In** is active, Soundscape will drop into Record mode at the Left Locator.

If **Auto Punch Out** is active, Soundscape will either toggle from Record mode into Play mode at the Right Locator, or stop if you have chosen that option in **Record Setup**.




If **Auto Punch Out** is not active, just click Stop on the Tape Transport (or press the ***Down Arrow*** or ***Space*** key on the computer keyboard) to stop recording.

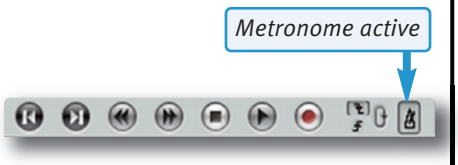
The track arming button will change from white to red whilst in Record mode (in the Record Track Column and in the corresponding track insert in the Mixer) and the Time Axis will appear shaded in the recording range. If you click Stop or press the ***Down Arrow*** key before the Right Locator is reached, the recorded Parts will be truncated to the Current Locator position.

The recorded Parts/Takes are automatically named after the mixer column that contains the corresponding track insert. You can change the Part names and Take names very easily later on if you wish (as described in the *Info tool* section of the *Editing Tools* section of this manual).

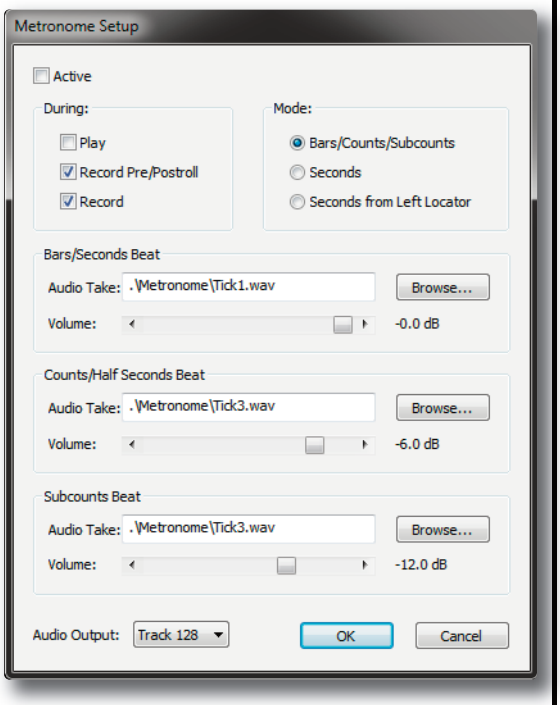
*NOTE: If the recording is interrupted (ie. if there is a power failure or PC crash), the recorded Takes will only be missing the last couple of seconds before the interruption.*

## Metronome

The Soundscape metronome is activated and deactivated by clicking on the metronome icon in the Tape Transport Bar.




To open the **Metronome Setup** window, select **Settings** / **Metronome Setup...**:




The **Active** box at the top of the display duplicates the activation icon in the Tape Transport Bar.

The three **During** options define when the metronome is played. You can also set the time **Mode** to **Bars/Counts/Subcounts**, **Seconds** (from the start of the Arrangement) or **Seconds from Left Locator**.

You then need to select which audio file is used for each element of the metronome, using the **Browse** buttons. You can then define the volume for each beat using the **Volume** sliders.

*Note that default metronome sounds are copied to your hard drive during Soundscape V7 installation.*

You also need to assign the metronome an audio track by clicking on the **Audio Output:** button at the base of the window, and selecting a track from the scrollable drop-down list which appears.

*Note that this track will be 'stolen' from the Arrangment whenever the metronome is active.*

You can now click **OK** to close the display, before performing one last element of the metronome setup: You must create a Track Insert in the Mixer to correspond with the output Track you have defined in the **Metronome Setup** window.

*See the Mix Window section of this Chapter for more information on creating Track Inserts.*

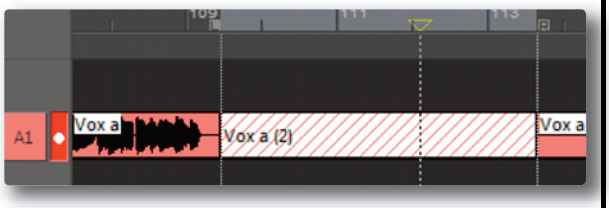
## Auto Punch In/Out

The Auto Punch In/Out recording mode allows new material to be recorded and inserted into existing Parts in the Arrangement. This is still non-destructive, but the Part is automatically replaced in the Arrangement (ie. rather than overwriting anything on Disk, the new Part is placed on Top).

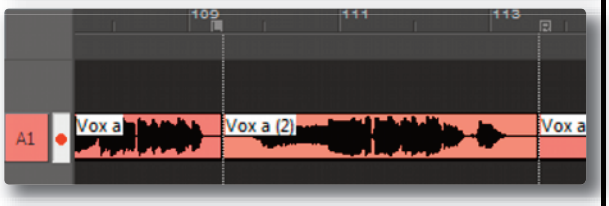
To auto punch in to an already recorded track, position the Locators, arm the corresponding record track and check that Auto Punch In/Out are active. Hit the Record button and the existing Part will automatically be cut at the Locator positions and muted in the Auto Punch In/Out time range, even if it is on a different virtual track. If you decide not to keep the new recording, click **Undo Record** in the **Edit** menu (or press ***Ctrl+Z*** on the computer keyboard).

The pre-existing audio is not deleted from disk when you record in **Auto Punch In/Out** mode, so even if the Punch In and Punch Out points (Locator positions) were not defined accurately, the edit point can be adjusted later.

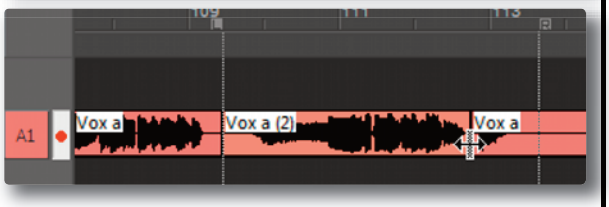
For example:
To the right, a section of a vocal Part recorded on track 1 needs to be replaced. The Locators are dropped in the area where the Auto Punch In is required.




The Recording will automatically Punch Out at the Right Locator. If you hit stop, the Waveform is calculated.

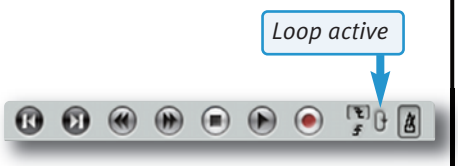



With the Trim Tool (**Include Adjacent Parts**) you can fine tune the in/out points to make the edit between original and overdub seamless.



## Loop Recording and Loop Stacks

'Loop' mode is activated by pressing the Loop button in the Tape Transport Bar.

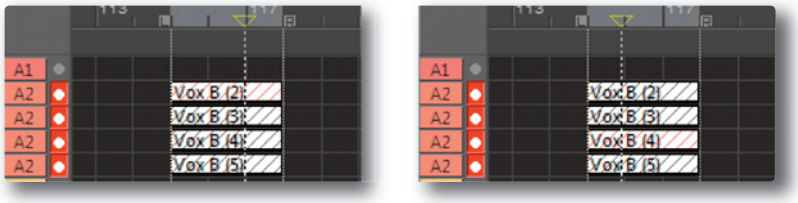



If recording is started with Loop mode and Auto Punch In/Out activated, Soundscape drops into recording when the Current Locator reaches the Left Locator, then starts again at the Left Locator when the Current Locator reaches the Right Locator (unless **Auto stop** is set to **at end of recording loop stack** in the **Settings** / **Record Setup** menu).

If the Stop button on the Tape Transport (or the ***Down Arrow*** key) is pressed just before the Right Locator, the last recorded Take is preserved. Otherwise it is overwritten during the next pass.

If recording is started with Loop mode active and with an armed record loop stack, recording is redirected to a new virtual track each time the Current Locator returns to the Left Locator.

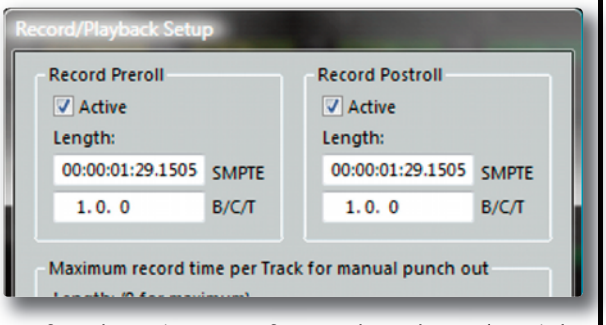
Up to nine Takes can be recorded in this way using a single record track without overwriting any of the first eight. After the last Part/Take in the record loop stack has been recorded, the same process is repeated, starting on the first virtual track of the record loop stack. From that point on however, the existing Parts/Takes in the stack are overwritten by any new ones. Recording can be stopped at any time.




*NOTES:*
*An existing record loop stack in the Record Track Column will only be taken into account if the Loop button is pressed.*
*If Soundscape is in Loop mode and the* ***Seamless loop during record/playback, if looping is active*** *option is disabled, there will be a short gap in playback when the Current Locator jumps from the Right back to the left Locator. This may be useful, if slaving a Sequencer via ASIO Positioning protocol where the sequencer needs to follow the loop.*

## Pre and Post Roll, Auto Stop

Clicking **Record Setup** in the **Settings** menu opens the **Record Setup** window, where a recording preroll and/or postroll can be defined and activated.




When recording is initiated with **Preroll** active (box checked), Soundscape will play from a position the specified preroll length before the Left Locator, and drop into Record mode when the Current Locator reaches the Left Locator. This is especially useful when a performer needs a cue during tracking.

If **Postroll** is active (box checked), Soundscape will continue to play after dropping out of Record mode at the Right Locator, for specified postroll time, before either:

- stopping (if **Auto stop** is set **At end of recording** and unless Loop mode is active);
- returning to the preroll position or Auto Punch In point (when recording a loop or loop stack) and starting a new cycle;
- stopping when the last track in a record loop stack has been recorded (if **Auto stop** is set **At end of recording loop stack**).

*NOTE: The preroll setting only has an effect if Auto Punch In is active, and the post-roll setting only has an effect if Auto Punch Out is active.*

## Recording while Slaving to external Timecode

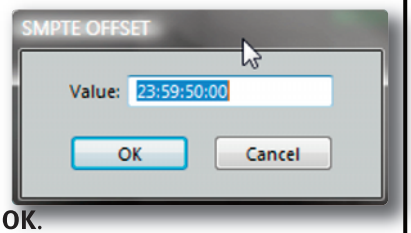
In any 'Synchronisation Slave' mode, whenever a Start or Jump message is received, Soundscape needs to fill its audio buffers with data from the disk before it can lock to the incoming Time Code. The time this takes depends on the number of tracks to be played back, and the speed of the disk and PC in general, etc. The value of this preparation time can be entered in the **Sync Setup** window under **Settings** / **Sync In**. The default is 2500ms, but any value between 100ms and 4000ms can be set. Since hard disk performance is always improving, extremely low settings are generally sufficient nowadays, especially for simple material.

However, when recording with Soundscape in Slave mode using Auto Punch In, in case the Time Code Master reaches the punch in point before Soundscape has had sufficient time to lock to the incoming Time Code, recording will simply not start. For this reason, playback from the Time Code Master device should be started before the punch in point, by a duration fractionally higher than the preparation time.

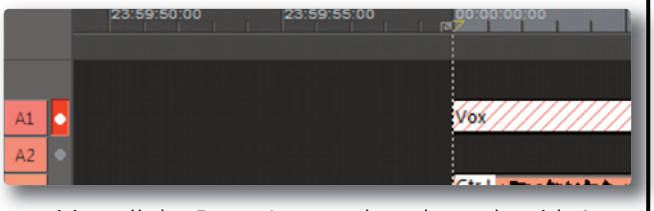
Conversely, if the preparation time is set too low for the conditions (eg., if the disk is too slow and a lot of tracks must be played back while recording), it can result in Current Locator 'jumps'. This is because each time the audio buffers are ready for starting playback at a targeted time position, the software detects that this position has already been passed and the whole process starts again.

The default setting is appropriate in most cases, as long as the Master device is not started too close to the punch in point. Another fail-safe solution is to keep Soundscape in Play mode, start the Time Code Master, and hit record when Soundscape's Current Locator is moving.

It is good practice to leave some empty time at the beginning of a recording in any device or system that can synchronise with another, whether as Master or Slave. This allows maximum flexibility for overdubs that require a count-in, or for starting an instrumental part on an upbeat before the 'beginning', etc., by giving the Time Code Slave the necessary time to lock.




However, even if you do this, you may be faced with situations where you need to start recording in Soundscape in a Time Code Slave mode with a punch in at SMPTE: 00:00:00:00 (Bar 1, Beat 1). This can happen if, for example, the Time Code Master device already contains audio data that must be played back from 00:00:00:00. In such cases, open the **SMPTE Offset** window by clicking **SMPTE Offset** under the **Tape** menu or the value in the SMPTE Offset readout, enter a value (eg., 23:59:50:00 for 10 seconds of empty time before 00:00:00:00 is reached in the Arrangement), and click **OK**.




Place a Locator at 00:00:00:00, or at any other position in the Arrangement where the first Part should start, select all the Parts in the Arrangement (by clicking **Select all Parts** under the **Global** menu or pressing ***Shift+A*** – this way you can be sure that all existing Parts are actually selected, including any Parts that are not currently visible), and use the 'Move To Locator' tool (as described in the *Editing Tools* section of this manual) to position all the Parts. In record ready mode with Auto Punch In made active (after you have pressed the track arming button), the Arrange window should look similar to that shown above (notice the values in the Time Axis).

Now set the Time Code Master device to use a similar offset, and use it to initiate playback. This will provide Soundscape with the necessary time (and more) for locking to the incoming Time Code before 00:00:00:00 is reached. Recording will then start at the punch in point (Left Locator) as normal.

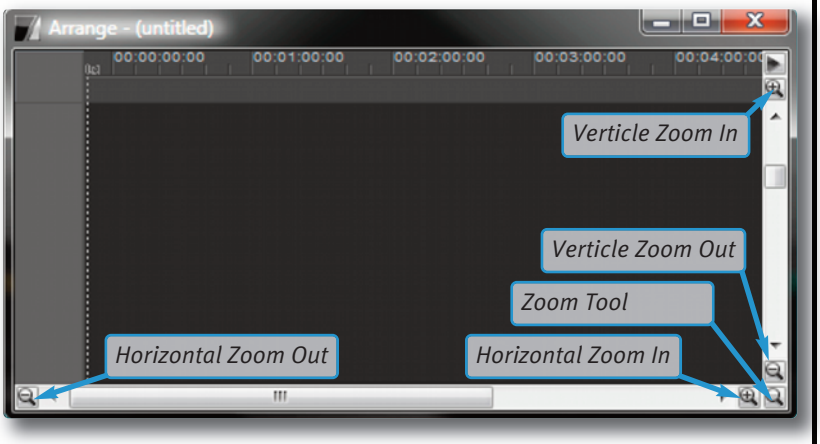
There might be a further difficulty if the Time Code Master device is not capable of sending Time Code before its 00:00:00:00 time position. In that case, you could try other solutions such as moving the existing audio in that device to a later time position, or slaving both Soundscape and the other device to a Time Code Master that is capable of providing such an offset (such as a MIDI interface with SMPTE/MTC generator etc...).

# Arrangement Navigation

## Zoom Overview

The zoom level for the Arrange window can be controlled in several ways:

- with the four Zoom buttons
- with the Zoom Tool
- using key commands
- using the Context Sensitive Edit tool
- with an external Hardware Controller supporting the MCU, HUI or JL Cooper CS/MCS Protocols via Console Manager.




## Using the Zoom buttons

Soundscape provides a wide variety of different Zoom Levels in both horizontal and vertical directions. At the highest Zoom Level, 1 Pixel on screen represents 1 Sample of the Audio file. Depending on the Display resolution, the Arrangement in the lowest Zoom Level can show several hours horizontally and all 256 Tracks vertically.

The waveform or automation data for each Part in the Arrange window is automatically displayed when the vertical and horizontal zoom levels are sufficient to display anything sensible.

*At the highest Zoom levels, the waveform data is extracted directly from the Take; In lower levels, the waveform data is read from a special Waveform Image File (.wvf extension) with a compressed resolution.*

### Horizontal Zoom

If the Current Locator is visible when zooming in horizontally, it is used as the centre of the zoom and remains in focus while the Arrangement 'expands' around it. Therefore, if you wish to zoom in on a particular point in the Arrangement, it can be helpful to place the Current Locator at or near that point.

If the Current Locator is not visible when zooming in horizontally, the visible centre of the Arrangement stays fixed while zooming in.

*It is often preferable to use the Zoom tool when accuracy is required. See the following page.*

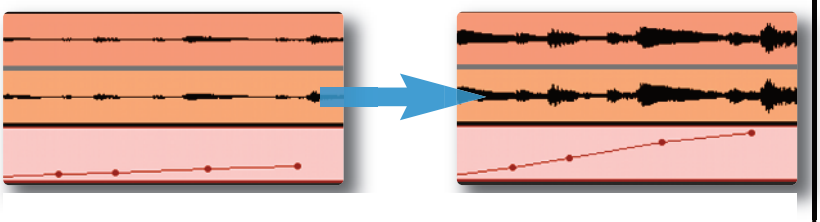
### Vertical Zoom

When zooming vertically, the top edge of the uppermost visible virtual track below the Time Axis stays fixed while the Arrangement is zoomed in or out.

If you need to zoom in to edit a particular Part, it is helpful to make sure that this Part is shown on the top virtual track before zooming in (this can be done by scrolling the window or moving the Part vertically).

### Waveform Zoom

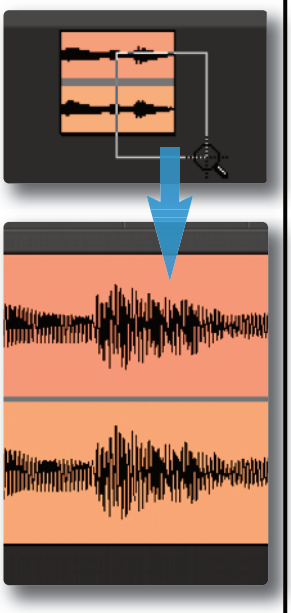
Right-clicking the Vertical Zoom In/Out buttons zooms only the Waveform or Automation Curves inside the Parts, increasing the vertical amplitude of the audio waveforms and automation curves. This does not alter the audio or automation data, but allows you to see low level waveforms and automation points, even in lower zoom levels. High amplitudes may be visually cut off at the top.



## Zoom Tool



Clicking once on the Zoom Tool (or pressing the ***Z*** key) will turn the mouse pointer into a magnifying glass when hovering over the Arrange Window. You can draw a rectangle in the Arrange window by holding down the left mouse button and dragging with the mouse. Releasing the mouse button will zoom into the area selected. The mouse pointer will then revert to its previous shape and function.



It is also possible to zoom in directly to sample level with a single click at the required position in the Arrange window (without drawing a rectangle). However this can be disconcerting, unless the zoom level is already very high. Coming from a low zoom level, it is impossible to reliably target an area of, for example, 300 samples.

The Zoom Tool can also be used to zoom out by drawing a selection box that extends beyond the current view of the Arrangement. The Arrange window will be scrolled automatically in the required direction, thus selecting a bigger section than currently visible.

## Zooming using key commands

Key commands are also available for zooming:

- Zoom In Vertical ***Alt+Up Arrow***
- Zoom Out Vertical ***Alt+Down Arrow***
- Zoom In Horizontal ***Alt+Right Arrow***
- Zoom Out Horizontal ***Alt+Left Arrow***

## Zoom and Position History

By double clicking the Zoom Tool (or hitting ***Z*** twice) you can quickly return to the previous view, which includes horizontal/vertical zoom level and horizontal/vertical position. This is an extremely useful way to switch between Project Overview and Track Detail View.

Soundscape remembers the last 8 views in the Arrange window, and their chronology. You can move either way through this view sequence by using the left and right mouse buttons when double-clicking the Zoom Tool. The left button is used to view previous (less recent) views, the right button is used to return to next (more recent) views. (Alternatively, you can press 2x ***Z*** and ***Shift+***2x ***Z*** keys instead.)

Views created using the Context Sensitive Edit tool are also remembered in the view sequence.

*NOTE: If you move back to an earlier view in the view sequence and then use the Zoom Tool again, the next view inside the sequence is replaced. For example if you go back to the fourth view and use the Zoom tool, the fifth view in the sequence is replaced by the new one.*

## Scrolling the Arrange window

The Arrange window can be scrolled using the Scroll buttons in the usual Windows way. It can also be scrolled using the following key commands:

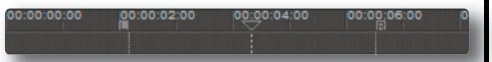
- Scroll Up ***Ctrl+Alt+Up Arrow***
- Scroll Down ***Ctrl+Alt+Down Arrow***
- Scroll Left ***Ctrl+Alt+Left Arrow***
- Scroll Right ***Ctrl+Alt+Right Arrow***

### Time Axis

The Time Axis, across the top of the Arrange window, shows time positions either in bars, counts and ticks (960 per quarter note)...



...or in SMPTE time as hours, minutes, seconds and frames




Toggling between the two types of time divisions can be done by selecting the corresponding option under the **Settings** / **Time Axis** menu.

When working with musical time divisions at a low zoom level, only some bar numbers are shown. At a high zoom level, finer time subdivisions are shown, down to single ticks.

When working with SMPTE time divisions at a low zoom level, hours are shown. At a high zoom level, each single frame is visible as a divider.

*NOTE: The snap value can be set to quarter, 10th, or 100th frame, and at maximum zoom level with the Time Axis set to SMPTE time, the mouse position readout shows single samples for ultimate precision while editing.*

## Arrangement Parts

Parts represent blocks of audio or automation data used in the current Arrangement. Each Part plays back data from a Take stored on disk. A Part can represent the full information of a Take or just a section.

Data from a single Take can be used by any number of Parts at any point in the Arrangement without requiring any additional disk space; Editing Parts in the Arrangement (eg., copying, cutting etc.) does not affect the corresponding Takes on disk.

Altering audio data by processing a Part (eg., normalising, pitch shifting, or reversing it) will result in the creation of a new Take of the same length as the processed Part (the length may be different from the original if, for example, the Part is time stretched/compressed). Parts have individual parameters, such as start point in the Arrangement, start point in the corresponding Take, and Length. Some of these parameters are specific to a Part type (eg., only Audio Parts have a Start and End Part Volume setting), and all the Part parameters can be displayed and edited using the Info tool as described in the *Editing Tools* section in this chapter.

Most editing and processing operations which can be performed on a single Part, can also be performed on a group of selected Parts. You can, for instance, copy or move a group of Parts, make a cut across several Parts at the same position, or delete several Parts at once.

Single or multiple Parts can be selected for editing by clicking in an empty space in the Arrange window and dragging the mouse pointer to draw a 'selection box':




Any Part which is touched by the selection box will be selected and will turn grey when the mouse button is released. The selection box can be started on a Part if the ***Shift*** key is held down while clicking (if the ***Shift*** key is not held down, clicking on a Part will trigger an editing or processing operation).



When certain tools are assigned to the mouse button being used, no selection box can be drawn. These tools are:

- Mixdown Tool
- Multitrack Scrub Tool
- Solo Scrub Tool
- Solo Tool

However, holding down ***Shift*** or ***Ctrl*** allows Part selection even when one of these tools are assigned to the mouse button.

If you need to draw a selection box beyond the currently visible section of the Arrangement, drag the outline of the selection box to any Arrange window border and the Arrangement will start to scroll.

A Part can be individually selected or deselected by clicking on it while holding down the ***Ctrl*** key on the computer keyboard.

Clicking in an empty space in the Arrange window, or performing an editing or processing operation on a non selected Part, will cancel all current Part selections.

The **Keep Part selection** item in **Settings** / **Preferences** / **Arrangement** determines whether the currently selected Part(s) will be automatically deselected when an editing or processing operation has been performed. When a number of editing and/or processing operations need to be carried out on the same Parts, it is useful to tick this option in order to avoid having to repeat the selection process over and over again.

# Locators and Markers

## Left and Right Locators



The Left and Right Locators are used to define a target area for Auto Punch In/Out recording, looping and loop recording; They also affect the operation of certain editing tools (eg., Mixdown tool, Crop tool...), are used by some of the Global functions (as described in the *Global Menu* Section of the *Menu Reference* chapter), and are used as snap points for certain snap settings. They can also be used for automation snapshot recording, etc.

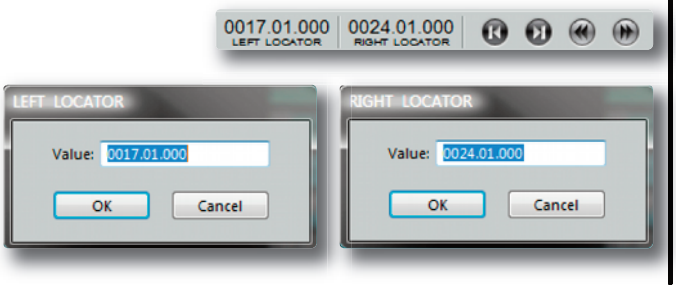
The time interval between the Locators is displayed in the **L<–>R Display** inside the **Status bar**.

## Setting the Left and Right Locators

To position the Left or Right Locator, click at the required position in the Time Axis with the corresponding mouse button. The Locator will be set to the mouse position, or to the closest snap point if the Snap function is active. A Locator can also be moved along the Time Axis by clicking and dragging its 'handle' with the corresponding mouse button.

While in play back or record, 'on the fly' positioning of the Locators is possible by using the ***L*** and ***R*** keys. In this case, the snap setting is ignored.

Alternatively, you can click the Left or Right Locator Display in the Tape Transport Bar to alter the time position in a dialogue box. Depending on the Time Axis setting, SMPTE Time or Bars/Beats/Ticks are displayed.




You can type ***Space*** instead of '**.**' or '**:**' and leading zeros are not required. The time positions entered in the examples to the right will appear in the displays as 00017.01.000 and 0024.01.000 .

## Current Locator

The Current Locator shows the current playback or Song Position the Arrangement.



Two Displays in the Tape Transport Bar show the Current Locator Position as **song position** in Bars/Beats/Ticks and as **SMPTE Time** in Timecode Format.




The Current Locator can be positioned in several ways:

- ...with the Tape Transport Buttons
- ...by clicking or right-clicking in any empty section of the Arrange window. If a tool assignment is prohibitive (ie. if the Multitrack Scrub tool is selected), pressing the ***Shift*** key will allow positioning of the Current Locator. This is not possible while in Play or Record mode.
- ...by clicking at the required position on the Time Axis while holding the ***Shift*** key. This is particularly useful when editing at a high zoom level and/or when there is no empty area visible in the Arrange window.
- ...by holding the ***Shift*** key and pressing the ***Left Arrow*** or ***Right Arrow*** key to move the Current Locator to the previous or next snap point (if the Snap function is active) or sample (if the Snap function is inactive).
- ...by holding the ***Ctrl*** key and pressing the ***Left Arrow*** or ***Right Arrow*** key to move the Current Locator to the previous or next Part beginning or end.
- ...by clicking the button at the bottom of the left or Right Locator Display, to move the Current Locator to the current position of the corresponding Locator. This function is not available in Play mode or Record mode:

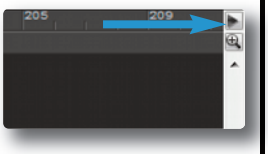



- ...by clicking the **Song Position** or **SMPTE Time** display in the Tape Transport Bar and entering a value in the dialogue box. This function is not available in Play or Record mode.

Pressing the ***C*** key on the keyboard will centre the Current Locator inside the Arrange window.

It is also possible to keep the Current Locator in the centre of the screen during playback or recording ('Tape Head' View) by selecting **Auto Scroll** in the **Settings** / **Preferences** / **Arrangement** menu.

By default the screen is redrawn automatically to follow the Current Locator, whenever it reaches the right side of the Arrangement. This behaviour can be disabled by clicking the 'Follow Current Locator' button located in the top right corner of the Arrange window (or by pressing the ***J*** key). The button toggles between a Play Icon (follow Current) and an illuminated Pause Icon (follow Current off).




Disabling 'Follow Current Locator' is especially useful while focussing on a particular edit point and working at a high zoom level.

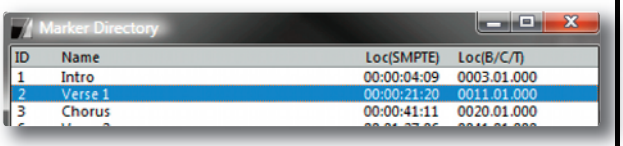
## Markers

The Marker Bar runs under the Time Axis in the Arrange window and can be shown or hidden using the **View** menu. You can insert up to 999 named Markers per Arrangement. Markers are saved in the Arrange file.




## Marker Directory

The Marker Directory can be shown or hidden by ticking or unticking **Marker Directory** under the **View** menu, or by pressing the ***W*** key. The window to the right shows a text listing of Markers, with their ID number, name and time position in both SMPTE and bars/counts/ticks.



## Inserting Markers

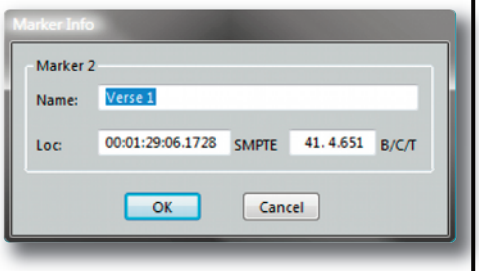
Markers can be inserted 'on the fly' at the current song position by pressing the ***Insert*** key (this works in stop, playback and recording modes).

Each time a Marker is inserted, its ID number is automatically incremented. However, if you type any ID number from 1 to 999 before pressing ***Insert***, the ID of subsequently inserted Markers will be incremented from that number, allowing markers to be used to identify different sections of an Arrangement (eg., Markers 100-120 for song section 1, Markers 200-220 for song section 2, etc.).

*NOTE: It is possible to edit Markers' names and positions while in Record or Play mode. The Marker Directory remains 'on top' when you insert Markers. This allows you, for example, to listen to a backing track during recording, and have dropped Markers with Notes by the time recording is finished.*

## Naming Markers

In the Marker Directory, double-clicking on a Marker entry opens the **Marker Info** dialogue box, where you can add or edit the Marker's name. The time position can also be altered (as **SMPTE** or **B/C/T** values).




Leading zeros are not required and you can type ***Space*** instead of '**:**' or '**.**'. The name can be up to 30 characters in length..

## Moving Markers

Any existing Marker can be moved to the current song position by typing its ID number and pressing the ***Enter*** key. Subsequent presses of the ***Enter*** key will then cause the next Markers (in the order of ID numbers) to move to the current song position. This process can continue until the last Marker is moved.

You can also use the mouse to move a Marker to a new position by left-clicking the Marker and dragging it. Active Snap settings will determine available Marker positions.

## Moving the Left and Right Locators to Marker locations

The Left and Right Locators can be moved to a Marker location by typing in the relevant Marker ID, then pressing the ***L*** or ***R*** key respectively. This can be done in Play or Stop mode, but not in Record mode.

## Moving the Current Locator to Marker locations

The Current Locator can be moved to a Marker location by typing in the relevant Marker ID, then pressing the ***Down Arrow*** key, or clicking the Stop button on the Tape Transport. This can be done in Play mode or Stop mode. In Record mode, the first Stop command will stop Soundscape, the second one will move the Current Locator as required.

Right-Clicking to the right of a Marker in the Marker Bar will move the Current Locator to that Marker's location.

In the Marker Directory, right-clicking on a Marker entry will move the Current Locator directly to that Marker's location.

## Playing from a Marker location

Playback can be started from a Marker location by typing in the Marker ID and pressing the ***Up Arrow*** key, or clicking the Play button on the Tape Transport. This can only be done in Stop mode.

## Deleting Markers in the Marker Directory

Markers can only be deleted in the Marker Directory. Click on an entry and press the ***Delete*** key.

Several Markers can be selected for deletion: holding the ***Shift*** key allows selection of a range of markers by clicking the first and last Markers of the required range, and holding the ***Ctrl*** key allows individual entries to be added to or removed from the selection by clicking on them.

## Tape Transport Bar

The Tape Transport Bar, located at the bottom of the main Soundscape Window, contains the above mentioned displays for Locator/**Song Position**, the Transport Section, and buttons for Auto Punch In/Out and Loop Playback/Record as well as displays for musical Time Signature (**Sign**), **Tempo** and **SMPTE Offset**.






**To Beginning**

Click this button or press ***Home*** to move the Current Locator to the beginning of the first Part(s) in the Arrangement.



**To End**

Click this button or press ***End*** to move the Current Locator to the end of the last Part(s) in the Arrangement.

If you hold down ***Shift*** and click these buttons, the Current Locator will move to the previous or next snap point if the Snap function is active, or to the previous or next sample if the Snap function is inactive. This is particularly useful when working with video and Snap set to Frames. This operation will interrupt playback or recording.

If you hold down the ***Ctrl*** key while clicking these buttons, the Current Locator will move to the previous Part beginning or next Part end. Using this function will interrupt playback or recording.



**Rewind and Fast Forward**

Press and hold these buttons to rewind or fast-forward the playhead – rewind and fast forward remain active as long as the buttons are held. If pressed during Play mode, playback will resume as soon as the button is released. The ***Left Arrow*** and ***Right Arrow*** keys duplicate these buttons.

*NOTE: The speed of Rewind and Fast Forward can be increased by activating **Fast REW / FF** in **Settings / Preferences / General**. In the same menu, **Scrub during REW / FF** can be activated, causing Reverse Play and Fast Forward to be performed at twice the playback speed.*



**Stop**

If Soundscape is in Play or Record mode, clicking the Stop button (or ***Down Arrow*** key) once will stop playback or interrupt the recording at the current song position.

Subsequent clicks on the Stop button will toggle the Current Locator between the Left Locator position and the start of the Arrangement. Clicking the Stop button after typing a Marker number on the numerical keypad positions the Current Locator to the corresponding Marker location.



**Play / Repeat Memory**

Clicking the Play button (or ***Up Arrow*** key) will initiate playback from the current song position. Clicking the Play button after typing a Marker number on the numerical keypad starts playback from that Marker location.



**Record**

Clicking the Record button (or the numeric **+** key) initiates recording from the current song position (if Auto Punch In and Pre-Rolls are not active).

Recording can be started while Soundscape is stopped (in which case the Play button is also engaged automatically), or during playback (manual punch in), and can be stopped by clicking the Record button again, or the Play button (manual punch out). Clicking any other tape transport button also stops recording. If the Auto Punch Out function has been activated, recording stops when the Current Locator reaches the Right Locator.

*NOTE: Recording behaviour depends on various settings (Auto Punch, Pre-/Post Roll, Loop Recording).*

## Using the computer keyboard for Tape Transport




### Play/Stop Toggle

The ***Space Bar*** toggles between Play and Stop Mode. When play is stopped with the ***Space Bar***, the Play Pointer remains at the Stop location.

The ***Up Arrow*** also toggles between Play and Stop. However, when play is stopped with the ***Up Arrow***, the Play Pointer returns to the previous Play location. This is particularly useful for repeated checking of a section of audio.

### Stop

The ***Down Arrow*** pressed once during play/record stops the current locator/finishes a recording. Pressed once in Stop Mode, it will place the Current Locator at the Left Locators Position. Pressed again, it will place the Current Locator at the Arrangements beginning. If the Current Locator is either at the Left Locator position or beginning of the Arrangement, the ***Down Arrow*** key will act as a toggle to jump between these two positions.

### REW/FF

The ***Left Arrow*** and ***Right Arrow*** keys can be used to control the Rewind and Fast Forward tape transport buttons. However, if these keys are used to initiate rewinding or fast forwarding, the Current Locator will keep moving in the chosen direction until a tape transport button is clicked (or until an equivalent key is pressed), or until it reaches the beginning or end or the Arrangement.

If **Scrub during REW / FF** is enabled (**Settings** / **Preferences** / **General**), pressing the ***Left Arrow*** will rewind in 1x playback speed, providing a 'Reverse Play' mode. Repeated key presses will increase REW speed incrementally, and subsequent key presses on the ***Right Arrow*** will decrease the REW speed incrementally until Playback flips forward and further key presses increase the FF playback speed.

### Auto-Punch and Loop

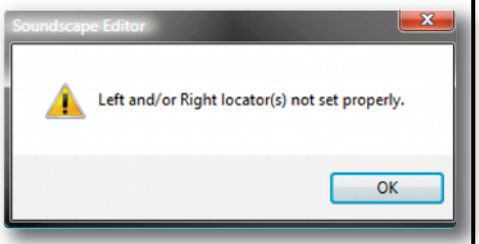
Clicking these buttons or typing the corresponding key commands toggles the Auto Punch In and Out modes and the Loop mode on or off.

**Auto Punch In** (***Ctrl+Alt+I***)

**Auto Punch Out** (***Ctrl+Alt+O***)

**Loop** (***Y***)

If an Auto Punch In/Out recording is attempted when the Right Locator is positioned to the left of the Left Locator, the warning shown right will appear. Click **OK** and change the Locator positions.

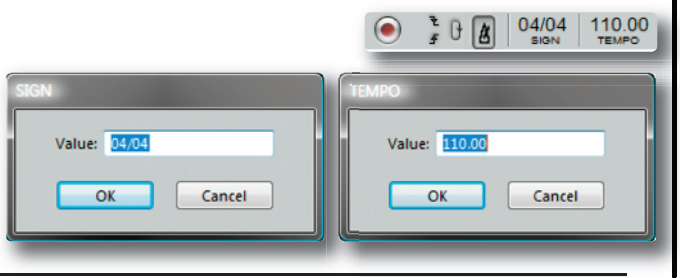



If Loop mode is made active in Play mode, when the Current Locator reaches the Right Locator it will return to the Left Locator. Soundscape will repeat the section between the two Locators until another tape transport button is clicked. If the Locators are set the wrong way around, playback will ignore the Loop mode.

*NOTE: If Soundscape is in Loop mode and the **Seamless loop during record/playback, if looping is active** option is enabled, Soundscape's Time Code Generator does not follow the loop. This is relevant when Soundscape is used as a Time Code Master. Please refer to the 'Record/Playback Setup' section of the Settings Menu chapter for details.*

## Metronome, Time Signature and Tempo

The metronome, time signature and tempo readouts are located at the right hand side of the Tape Transport buttons.




Clicking on the metronome icon to the left of the **SIGN** readout will make the metronome active ('pressed down') or inactive (not 'pressed down').

*NOTE: The metronome is configured via **Settings** / **Metronome Setup....** See the 'Settings Menu' section of this User Guide for more information.*

Clicking either the time signature (**SIGN**) or **TEMPO** readout opens a dialogue box in which their value can be entered.

When entering the time signature, the 'beats per bar' (first number) can be set to any value from 1 to 15 and the 'beat value' (second number) can be 4, 8 or 16. If you try to enter a value outside these ranges, the closest available value will be used instead. Leading zeros are not required and you can type ***Space*** instead of '**/**'.

The tempo can be set to any value between 30 and 240 beats per minute, with up to two decimal digits (eg. 170.86 bpm). The leading zero (for a tempo under 100 bpm) is not required and you can type ***Space*** instead of '**.**'.

The time signature and tempo in Soundscape is especially important while editing with musical Snap values or working in Sync with a MIDI Sequencer. Note that changing time signature and tempo does not change the absolute time locations or Playback speed of any part inside the Arrangement.

*NOTE: The Time Signature and Tempo cannot be edited when a Tempo Map is loaded (see Menu Reference Chapter).*

## SMPTE Offset

Clicking the **SMPTE offset** readout to the right of the Tape Transport calls up a dialogue box in which the required **SMPTE offset** value can be entered in SMPTE format. You can type ***Space*** instead of '**:**' and leading zeros are not required.




The SMPTE Offset can be used to align Soundscape correctly with time code from an external device. For example, if a section of film which requires an audio track has time code starting at 00:18:23:06 (0 hours, 18 minutes, 23 seconds and 6 frames), entering this Offset value will make the Arrangement also start at this time. In 'Slave' mode the Offset is subtracted from the external time code, and in 'Master' mode it is added to Soundscape's timing before the MIDI Time Code is generated.

The Offset parameter can also be useful when it is necessary to record from time position 00:00:00:00 in 'Slave' mode. Please refer to the *Recording while slaving to Time Code* subsection of this chapter for details.

Any value within the 24 hour range can be entered for SMPTE Offset.

# Editing Tools

***IMPORTANT:*** *The operation of the editing tools is described in detail in this chapter, and it is assumed that you are already familiar with the tool selection process as explained in the Menus and Toolbars section of this chapter. Once a tool has been assigned to a mouse button, it can be used by clicking with that mouse button on the Part(s), or in a few cases (eg., the Mixdown tool) by clicking in an empty space in the Arrange window. The* ***Alt*** *key can also be held down to assign one extra tool to each mouse button.*

*NOTE: The editing operations described for one single Part can often be performed identically on a group of selected Parts. Any significant differences will be highlighted.*



**Track Assign tool**

The Track Assign tool works on audio and automation Parts, and allows you to select which physical audio or automation track and which Part (or group of selected Parts) are assigned to each other. Track Assign can be activated with the ***Ctrl+1*** keys.

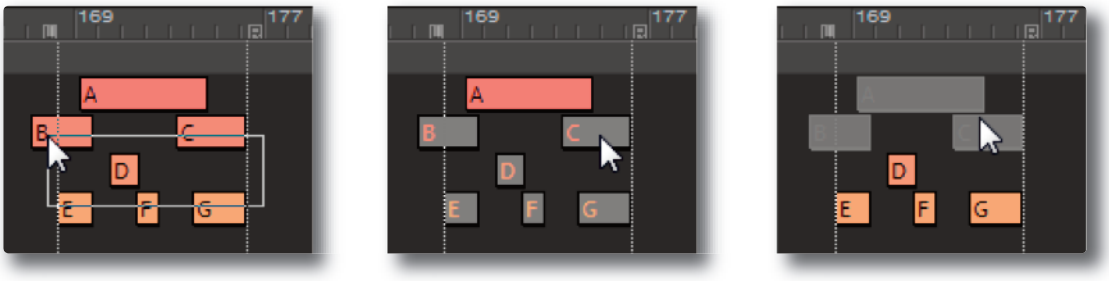
When you click on a Part with the Track Assign tool, a menu appears which lists all the selectable tracks in the appropriate category (audio or automation). To select a track, place the mouse pointer over its number before releasing the mouse button. If the tool is used on stereo or multi-channel linked audio Parts, the appropriate track pair or group which includes the chosen number will be selected. If the tool is used on a muted Part, this Part becomes active.

The effect of the Track Assign tool is dependent on the setting chosen for the **Multiple Active Parts with same Output and Location (for new edits)** item under the **Settings** / **Preferences** / **Arrangement** menu:

- If **Not allowed** is selected, when an active Part is assigned to a new track, any other active Parts that are already assigned to the chosen track and that are (even partially) in the same time range (and therefore overlap) are automatically muted.
- If the Track Assign tool is used on a group of selected active Parts of which several are in the same time range, the selected Parts all get the chosen track assignment, but only one of the Parts will remain active; The others are automatically muted, along with any other Part in the same time range which already had the chosen track assignment.

In the example below:

- Part A is assigned to track 7, Parts B and C are assigned to track 8, Part D is assigned to track 9, and Parts E, F, and G are assigned to track 10. Parts B, C, D, E, F, and G are selected by drawing a selection box (as a result they appear grey in the second image). The Track Assign tool is then used to assign them all to track 7.
- Part B is muted because it extends into Part E's time range, and Part C is muted because it extends into Part G's time range. Part A, which was already assigned to track 7 and which was not selected, is muted because its time range also includes Parts E, F, and G.
- Parts E, D, F, and G remain active.




If overlaps are allowed and **Topmost Part (portion)** is selected, this automatic muting does not occur.

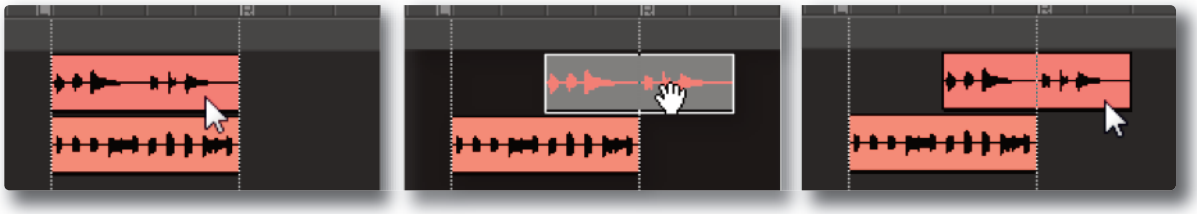
*Please also read the Mute tool section of this chapter and the 'Multiple Active Parts with same Output and Location (for new edits)' subsection of the 'Settings Menu' section of Chapter 5 for complementary information.*

## Move tool

The Move tool works on audio and automation Parts, and allows any Part (or group of selected Parts) to be moved to any position in an Arrangement. The Move tool can be activated with the ***Ctrl+M*** keys.

**To move a single Part**

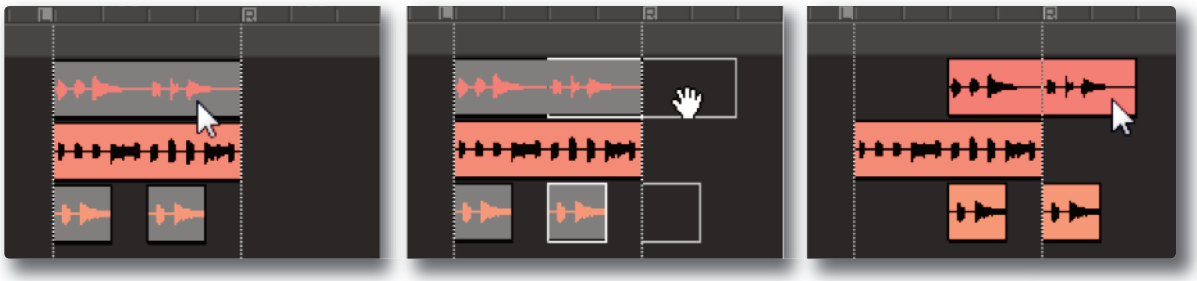
Once you have assigned the tool to a mouse button, click and hold any Part and drag it to the desired position. If the target position is outside the currently visible section of the Arrange window, hold the dragged Part on the window border and the Arrangement will auto-scroll in the corresponding direction. Release the mouse button to drop the Part at the mouse position. If Snap is active, the beginning of the Part (or its end, depending on the current snap value) will jump to the nearest snap point:



*NOTE: The tool also takes embedded snap points into account, as described in the 'Snap Point Edit tool' section of this chapter.*

**To move a group of selected Parts**

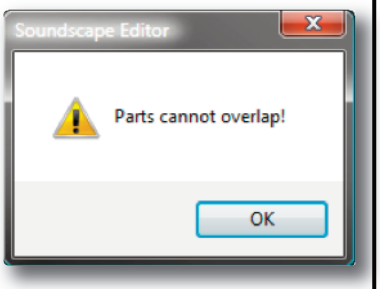
Click and hold any Part within a group of selected Parts to drag the whole group to a new location. If Snap is active, the Part you initially clicked on will jump to the nearest snap point when the mouse button is released, but the relative timings of the group will remain unchanged.



*NOTE: When an active Part is moved to a virtual track that has a record track assignment, the moved Part's own track assignment will be changed if necessary to match the record track if the **Adopt virtual track Output assignment when moving/copying Parts** option is activated under the **Settings** / **Preferences** / **Arrangement** menu. This applies whether the Part is moved individually or within a group.*

If the moved Part or group overlaps with an existing Part when you release the mouse button, several things might happen:

- If the **Snap** function is active, and **Not allowed** is selected for **Overlapping Parts**, the warning on the right hand side appears and the move is cancelled.
- If the **Snap** function is inactive, and **Not allowed** is selected for **Overlapping Parts**, Soundscape automatically puts the moved Part or group of Parts to the beginning or end of the overlapped Part, whichever is closer.
- If **Allowed** is selected for **Overlapping Parts**, the moved Part will simply be dropped at the current position when you release the mouse button, or jump to the nearest snap point if the **Snap** function is active, regardless of the overlap.




**Nudge**

It is possible to 'nudge' a Part (or group of selected Parts) by clicking and holding it with the Move tool and using the four arrow keys. The Part is moved vertically by one virtual track each time the ***Up Arrow*** or ***Down Arrow*** key is pressed. It is also moved horizontally to the previous or next snap point each time the ***Left Arrow*** or ***Right Arrow*** key is pressed If the snap setting is active, or to the previous or next sample if the snap setting is inactive.

## Copy tool

The Copy tool works on audio and automation Parts and allows any Part (or group of selected Parts) to be copied to any position in the Arrange window. The operation of the Copy tool is similar to that of the Move tool, except that the original Part or group will be left unchanged. The copied Part or group will be identical to the original and reference the same Take, so no additional Disk space is required for the copy. The Copy tool can be activated with the ***Ctrl+C*** keys.

**Nudge copy**

The copy of a Part can be 'nudged' while the original stays in place, using the Copy tool and the four arrow keys in the manner described for the Move tool above.

## Move & Copy Vertical tools

These tools works on audio and automation Parts. They only allow edits in the vertical direction (ie., from one virtual track to another). This ensures that no change in timing occurs. These can be activated with ***Ctrl+Shift+M*** or ***Ctrl+Shift+C***.

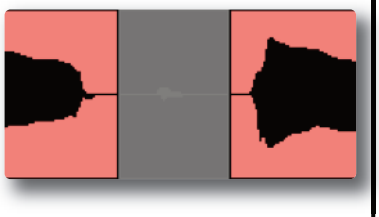
## Cut tool

The Cut tool works on audio and automation Parts, and allows non-destructive editing of any Part in the Arrange window. To cut a Part, click on it at the time position where the edit is required. If Snap is active, the position of the cut will correspond to the nearest snap point. The resulting two Parts will have identical volume, fade, and track assignment settings (although any existing fade curve across the two Parts will be displayed and remain active until you use the Fade In, Fade Out, or Volume And Fade Trim tool on one of the Parts).

The resulting new Parts can then be treated in exactly the same way as any other Parts, and can be copied or moved. For example, the Cut tool is useful for defining a region that requires a different volume or fade setting.

You can make cuts across several virtual tracks at once by selecting a group, and then clicking on one Part in the group. Any other selected Parts at the same timing position will be cut at the same point.

For precise editing, it is often necessary to deactivate Snap and zoom in to waveform level. For instance, to remove a click from a Part, the waveform can be cut at the start and end of the click, and the new Part that contains the click can then be deleted, have its volume reduced, or be muted (as illustrated right).



The Cut tool can be activated with the ***Ctrl+X*** keys.

*NOTE: You can also cut several selected Parts at the same time by clicking on one of them at the time position where the edit is required.*

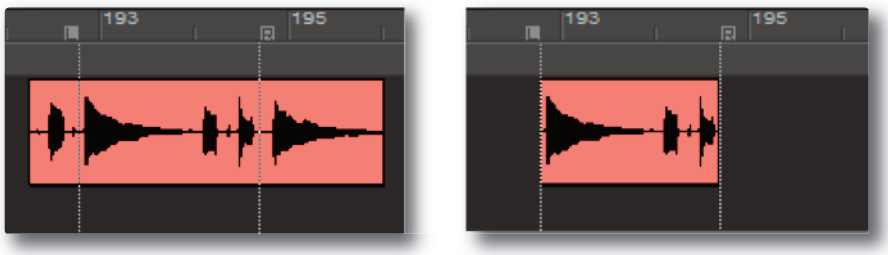
## Move To, Copy To & Cut At Locator tools

The Move to-, Copy to-, and Cut at-Nearest Locator tools work on audio and automation Parts, and allow Locator-based editing. The Current Locator, Left Locator or Right Locator (whichever the mouse pointer is closest to) determine the position of a Move, Copy, or Cut operation. With the Move tool or Copy tool, the beginning or end of the selected Part(s) will snap to the nearest Locator. This allows, for instance, sound effects to start or end at a time code reference which is set while playing, or by entering the Locator Position as a value in a Tape Transport bar readout.

These tools can be activated with the ***Ctrl+Alt+M***, ***Ctrl+Alt+C*** and ***Ctrl+Alt+X*** keys.

## Crop tool

The Crop tool works on audio and automation Parts, and reduces the size of a Part or group of selected Parts to the area defined by the Left and Right Locators. Any sections of the Part(s) outside that area therefore disappears from the Arrangement. The Crop tool can be activated with the ***Ctrl+3*** keys.




## Solo tool

The Solo tool only works on audio Parts and can be used to play individual Parts (or stereo/multi-linked Parts). To solo an active (not muted) Part, click on (or in front of) it and hold the mouse button down. The Part will be played from the mouse position, for as long as the button is held down. When the Part ends, the next active Part with the same track assignment will be played, unless it is on a different virtual track. If the mouse button is released and then pressed again without moving the mouse pointer, the same section of the Part will be repeated, making it easy to listen to a vocal phrase, identify a note, or generally check any detail in the audio.

The Solo tool can be used to play muted Parts. In that case, the track assignment used for playback is the one selected under **Settings** / **Muted Parts** / **Audio Takes Solo Output**.

For muted stereo/multi linked Parts, the corresponding track and following track(s) above the selected Solo Output will be used.

The Solo tool can be activated with the ***Ctrl+H*** keys.

## Solo Part Begin/End tool

The Solo Part Begin/End tool only works on audio Parts. It allows soloing of a single Part from beginning to end. If the mouse pointer is in the left half of the Part, soloing will start at the beginning of the Part and will continue until the end of the Part or until the mouse button is released. If the mouse pointer is in the right half of the Part, then soloing will start at the current mouse pointer position and stop at the end.

In addition, if the Left and/or Right Locator(s) are placed within a Part, you can solo up to a Locator, from a Locator, or from one Locator to the other. This allows edits to be rehearsed before cutting or deleting any section of the Part. Once rehearsed, the edits can be made at the Locator positions using the Move To, Copy To and Cut At Locator tools described earlier.

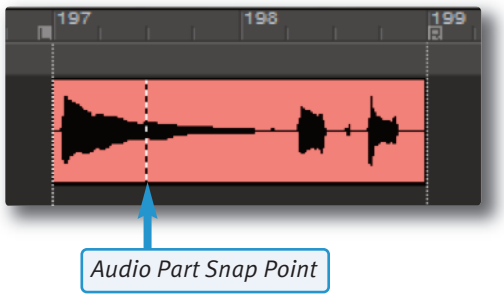
The tool also takes embedded snap points into account, as described in the *Snap Point Edit* tool section below.

The Solo Part Begin/End tool can be used to play muted Parts. The track used for playback is the one selected under the **Settings** / **Muted Parts** / **Audio Takes Solo Output** / **Unit x** menu. For muted stereo linked Parts, the corresponding odd/even numbered pair of tracks will be selected.

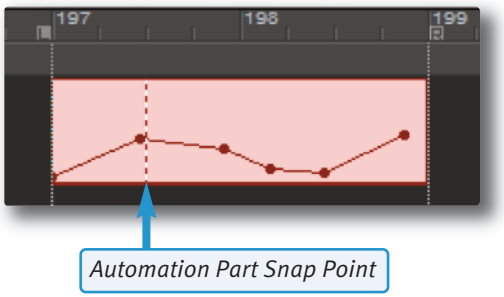
This tool can be activated with the ***Ctrl+Shift+H*** keys.

## Snap Point Edit tool

This tool works on audio and automation Parts. Clicking an audio or automation Part with The Snap Point Edit tool will embed a snap point within that Part, at the time position where the mouse pointer is located (or at the nearest snap point if Snap is active). ***Ctrl+Shift+D*** also activates this function. The embedded snap point appears as a dotted line across the Part:




*Audio Part Snap Point*




*Automation Part Snap Point*

Clicking on a Part which already contains a snap point will move that snap point to the new position (or to the nearest snap point if Snap is active).

Double-clicking a Part which already contains a snap point will delete that snap point.

Typically, an embedded snap point would be used to mark a point in an audio Take or automation curve which needs to be positioned precisely, eg., on a particular frame when working with video, or on a particular beat when editing music.

- If **Snap** is active, when a Part that contains a snap point is edited using the Move, Move to Locator, Copy, or Copy to Locator tool, that embedded snap point (rather than the Part's beginning) responds to the external snap points defined by the current snap setting. The embedded snap point is also taken into account when the **Move selected Parts to L locator** or **Copy selected Parts to L locator** function in the **Global** menu is used.
- If **Snap** is active and set to **Part** or **Prt+Mrk+L**, any snap point embedded in a Part (as well as its beginning and end) affects edits to other Parts performed using the Move, Copy, or Trim tools (if these edits are 'within range').

The Solo Part Begin/End tool will also respond to an embedded snap point, treating it as a Part beginning or Locator to start playback from, or like a Part End or Locator to stop at, according to the position of the mouse pointer.




The position of an embedded snap point can be checked or edited in the **Part Info** window, which is opened by clicking the Part with the Info tool. The snap point's position is displayed in **SMPTE** time and **B/C/T** counted from the Part's beginning.

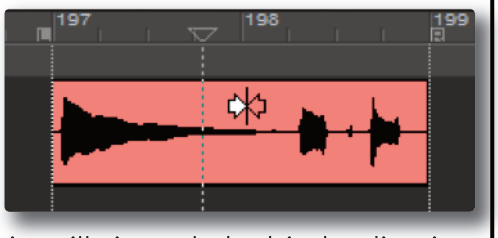
When editing the value using the computer keyboard, leading zeros are not required, and you can type ***Space*** instead of '**:**' or '**.**'

*NOTE: The embedded snap point is not saved with the Take but only with the Part in the Arrangement. Therefore, the same Take can be used by several Parts, with different snap point locations, in any number of Arrangements.*



## Scrub tools

These tools works on audio and automation Parts. The Solo (white) and Multitrack (grey) Scrub tools allow the audio to be played forwards or backwards, using the mouse to determine the playback speed and direction. These tools are also available using ***Ctrl+B*** and ***Ctrl+Shift+B***.




With the Solo Scrub tool selected, holding the mouse button down on an audio Part (or pair of stereo/multi-channel linked Parts) will cause the mouse pointer to change to a scrub icon.

The scrub icon consists of two opposing white arrows and a center line. When the Current Locator is moving, the arrow which points in the opposite direction to playback becomes transparent. Initially, the Current Locator will be aligned with the scrub icon's centre line. Dragging the mouse in either direction will trigger playback in that direction.

The selection made under the **Settings** / **Preferences** / **Arrangement** / **Scrub tools behaviour** menu determines how the Scrub tools react to further mouse movements:

- If **Locator based scrub** is selected, the distance between the scrub icon and the Current Locator determines the scrubbing speed, from zero up to normal speed. Scrubbing slows down as the Current Locator nears the scrub icon, and stops when it becomes aligned with the scrub icon's center line.
- If **Speed based scrub** is selected, the initial scrubbing speed is determined by the speed at which the Scrub tool is first moved. Further mouse movements in the same direction increase the scrubbing speed (which can exceed normal speed). Mouse movements in the opposite direction pull the Current Locator back, decreasing the scrubbing speed or changing the scrubbing direction.

*NOTE: In both cases, a higher zoom level allows slower scrubbing speeds.*

When the end of the Part is reached, the next active Part with the same track assignment will be scrubbed, unless it is on a different virtual track and **Topmost Part (portion) is played** is selected for the **Multiple Active Parts with same Output and Location** option (**Settings** / **Preferences** / **Arrangement**).

The Multitrack Scrub tool works in the same way as the Solo Scrub tool, except that it scrubs all active Parts. It is not necessary to point at a particular Part for playback to occur.

The Solo Scrub tool can be used to scrub muted Parts. In this case, the track used for playback is the one defined for soloing muted Parts (**Settings** / **Muted Parts** / **Audio Takes Solo Output**).

*NOTE: You can drop the Left and Right Locators while scrubbing, using the **L** and **R** keys.*

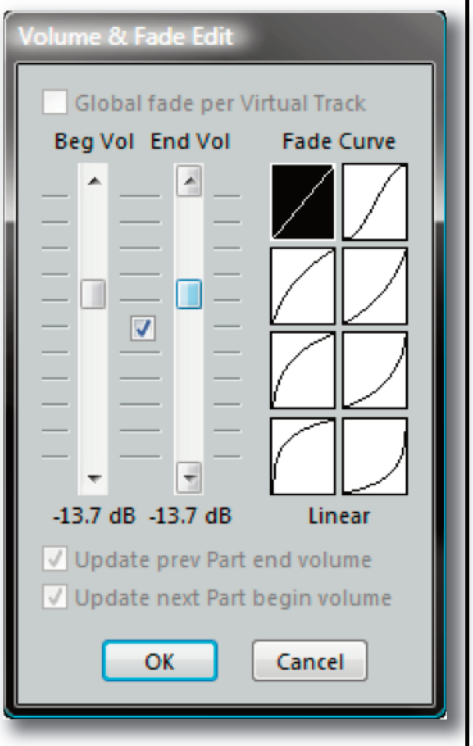


## Part Volume and Fade tools

These tools only work on audio Parts. Each Part in the Arrangement has a beginning volume, an end volume, and one of eight possible fade curve settings, and there are four tools which are used to edit these parameters. This provides an extremely powerful and flexible alternative to mix automation. If the **Show Part Volumes** option is enabled (**Settings** / **Preferences** / **Arrangement**), Part volume is indicated on-screen by the vertical amplitude of the coloured section of a Part. However, due to screen resolutions, volumes are only displayed from the third vertical zoom level on.

## Volume & Fade Edit tool

Select the tool and click on an audio Part. The **Volume & Fade Edit** window will appear (Also activated using ***Ctrl+V***):




The faders for beginning volume and end volume can be linked by checking the box between the two faders. In this way, the overall level can be changed without affecting the volume contour. When linked, the difference in dB between the volume settings is maintained. However, because the faders follow an audio taper law for dB versus on-screen movement, the distance between the faders on-screen will not remain constant as the faders are moved.

To edit the beginning volume or end volume individually, ensure that the check box between the faders is not selected.

You can select a fader either by clicking on it or using the ***Tab*** or ***Shift+Tab*** key(s). The selected fader will flash and can be moved either by clicking and dragging it with the mouse, or using the arrow keys (for a slow/fine level change), or using the ***Page Up***/***Page Down*** keys (for a fast/coarse level change). When using the arrow keys, ***Left Arrow*** and ***Right Arrow*** can function as ***Up Arrow*** and ***Down Arrow*** respectively.

The two check boxes, **Update prev Part end volume** and **Update next Part begin volume**, can be used if there is an adjacent Part before or after the edited Part and on the same virtual track. If a box is checked, the beginning or end volume of the corresponding Part will be automatically adjusted to match the volume of the edited Part, ensuring a smooth overall volume contour. Either check box will be dimmed if there is no corresponding adjacent Part.

*NOTE: The check box settings are not saved for each Part, but remain as you leave them each time you use the tool.*

There are 8 different fade curves providing one linear, one cosine, three exponential, and three logarithmic shapes. Each curve can be used for fading in or out. The currently selected curve setting is shown in black, and clicking on another curve will select it in blue, showing that a change has been made. The curves apply to a fade from one level to another, as well as to a fade in from silence (-96dB) or a fade out into silence.

*NOTES:*
*A linear fade curve has the effect of reducing the volume by a fixed amount per second. This means that after half of the fade time, the level is half of the original (-6dB), and after 3/4 of the fade time, the level is 1/4 of the original (-12dB), etc, leaving 1/4 of the time for the rest of the fade (to -96dB). This is a linear fade and has the characteristic that the level decreases very fast towards the end of the fade. An exponential fade, where the fade rate is faster at the start and more gentle towards the end of the fade will produce a more natural sounding fade out.*
*The maximum fade time for one single Part is 3 min 10 sec 05 frames (at 44100Hz). For fades longer than this, cut the Part into several smaller ones, select them all, and use the Fade In/Out tools, or use the global feature within the Volume tool to define a fade across all Parts.*

## Fade In & Fade Out tools

The Fade In and Fade Out tools change the beginning or end volume to -96dB respectively without affecting the other volume setting. The curve setting is left unchanged. They are also available via ***Ctrl+E*** and ***Ctrl+F***.

If you click with either Fade tool on a Part which already has a fade selected, the fade will be removed, and the value of whichever volume was highest will be applied to both. This is also the case when multiple Parts are selected, so that turning off the fade for multiple Parts will set all Part volumes to the highest volume found for all selected Parts. In other words, you will lose any volume contours that may have been previously defined!

## Volume Trim tool

The Volume Trim tool allows Part volumes to be edited in the Arrange window using the mouse. It provides a more direct approach than using the Volume & Fade Edit tool, yet both tools complement each other (ie., a volume curve defined using the Volume Trim tool will have the shape selected in the Volume & Fade Edit window). The amount of gain reduction being introduced is displayed beside the mouse pointer. Volume Trim is also available using ***Ctrl+Shift+V***.

**Basic operation:**

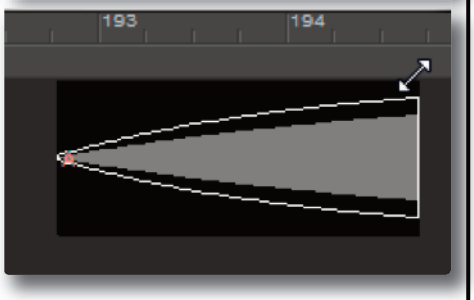
Click and drag the mouse up and down in the centre of a Part to change both its beginning and end volumes, leaving the slope unchanged.




Click and drag on the left side of a Part to change the beginning volume.




Click and drag on the right side of a Part to change the end volume.




Using the Volume Trim tool within a group of selected Parts will modify the Part clicked on and the previous and next Parts. When you click and drag in the centre of the Part, because both the beginning and end volumes are modified to maintain the fade shape, both the end volume of the previous Part and the beginning volume of the next Part are updated automatically. Clicking on the cut between two selected Parts will modify the end volume of the first Part and the beginning volume of the second. The volume of any selected Part on a different virtual track will be changed only if its beginning and end points are in the same timing positions as those of the Part being edited.

To easily define the volume contour for an entire virtual track, first enable **Keep Part selection** (**Settings** / **Preferences** / **Arrangement** or ***Shift+P***), then select Parts on the relevant virtual track as required and use the Volume Trim, Cut and Trim tools to modify the volume contour. If two adjacent Parts have different end and beginning volumes at their point of contact, causing a discontinuity which you would like to smooth out, drag each volume to the maximum level, release the mouse and then drag both volumes to the required level.



## Part Volume & Fades for multiple Parts

The four Volume and Fade tools can also be used if multiple Parts are selected, across one or several virtual tracks. The following paragraphs describe Volume and Fade tool operation in such circumstances.

**Volume & Fade Edit window**

With multiple Parts selected, the beginning and end volumes shown when using the Volume & Fade Edit tool are the beginning volume of the first selected Part and the end volume of the last selected Part, for the virtual track of the Part that you click on. This allows a group of Parts to be treated as one if required, with one curve and two volume settings.

- If the **Global fade per Virtual Track** box is not checked: Changes made to any parameter will be made for all selected Parts. For example, if the end volume is increased by 5dB, all Parts will have their end volume setting increased by 5dB. If the fade curve shape is changed, then all Parts will have the same fade curve shape selected.
- If the **Global fade per Virtual Track** box is checked: If the fade curve shape is changed, a smooth fade will be made across all selected Parts on each virtual track, starting at the beginning volume level and finishing at the end volume level, approximated to the chosen curve (shown in black). Each individual Part will then have this curve selected, allowing a closer overall approximation to the curve by selecting all the Parts again, using the Volume & Fade Edit tool, then turning off the global check box and selecting the linear curve.

If you use the Volume & Fade Edit tool for a group of Parts which have different curves, then the curve will not be shown in black, and you will not be able to alter beginning and end volumes separately; The box linking the faders is automatically checked and it cannot be unchecked. However, If you select a new fade curve shape, you will be allowed to use the fader link and **Global fade per Virtual Track** check box settings if desired. Note that all Parts will then have their curve set the same.

**Changing the overall level for multiple Parts**

If you want to change the overall level of a track which has contours already defined, use the Volume & Fade Edit tool, making sure that the **Global fade per Virtual Track** check box is unchecked and that the fader link box is checked so that the same changes are made to both beginning and end volumes of all selected Parts.

Alternatively, you can use the Part Info tool and click on one of the selected Parts which does allow the necessary level change to be made. Link the faders and change the level as required. All Parts will have their beginning and end volumes changed by the same amount.

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*NOTE: If some selected Parts have levels that are already too close to minimum or maximum for the same amount of change to be made, these will saturate at minimum or maximum volume. Sometimes the faders cannot be moved enough in the required direction, maybe because the beginning volume of the first Part is at minimum (eg., a fade in) and the end volume of the last Part is also at minimum (eg., a fade out). In this instance, both faders would be at -96dB and so the level of the track could not be reduced by, say, 6dB. The easy way to get around this is to select all required Parts except the fades in/out at either end. Make sure that the* ***Update prev/next Part end/begin volume*** *boxes are checked, and change the level. You can also use this technique to increase the level when there is a fade at either end, because the fade in and fade out will still be from -96dB.*

---

**Fading multiple Parts**

When you use the Fade In or Fade Out tool to fade across multiple Parts, each Part will take on the curve setting of the Part that you click on. Also, the shape for the entire fade will approximate to that curve so that the beginning and end volumes of each Part are selected automatically, according to points on the curve.

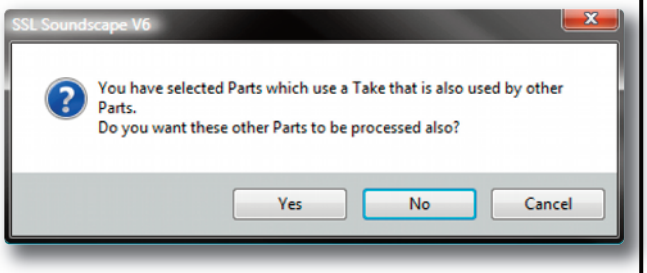
However, unless the edited Part has the linear curve shape selected first, the approximation to the overall shape required may not be as regular as expected, because the overall result will be, by definition, a succession of non-linear curves. To change the curve setting of each Part to linear, first select all the Parts, then use the Volume & Fade Edit tool and select the linear curve in the **Volume & Fade Edit** window, making sure that the global fade per virtual track box is unchecked.

## Normalize Process tool

This tool only works on audio Parts. Normalising can be used to adjust the overall level of a Part or the level of a whole Take so that its peak matches a value of your choice. This process is used typically to boost a signal which was not recorded at a sufficient level. A new Take is generated to disk and renamed automatically (eg., #1 or #2 is appended to the original Take's name). This tool can also be accessed via ***Ctrl+N***.

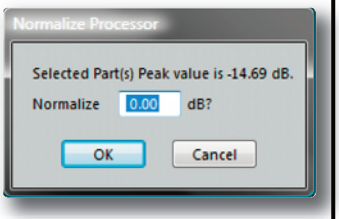
First, select a Part or group of Parts, and click with the Normalise tool.

If you have selected any Part which is not the entire length of the Take on the Disk, or which references a Take also referenced by other Parts which you did not select, the message shown right will be displayed.



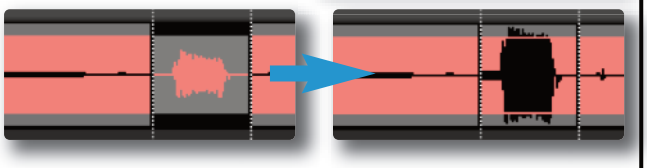

Click **No** to generate a new normalised Take just for the Part(s) selected, ie., the same length as the Part(s). Click **Yes** to normalise the entire Take and generate a new Take which is the same length as the original.

The Part(s) will then be scanned, and the highest peak value found for all selected Parts will be reported as shown right.




If you wish to continue with the normalise process, enter the required value for the new peak level and click **OK**, otherwise click **Cancel** to leave everything unaltered. Remember to include a minus sign when entering the value; Any positive value will be treated as 0dB.

If you go ahead, each selected Part will be increased in level by an amount which causes the reported peak level to reach the value you entered.



*TIP: You can use the Normalise function to determine noise gate 'on level' settings: Make a cut at each end of an area that you wish to 'gate out', then click on the resulting Part with the Normalise tool. It will show you the peak level detected for the selected Part. Click* ***Cancel*** *to avoid normalising the Part, then glue the cut section back into the main Part. If you use the Normalise value as the 'off level' value for the noise gate, it will mute all sections of the Part below that level.*

## Time Stretch/Compress Process tool

The Time Stretch/Compress Process tool (***Ctrl+Shift+T***) only works on one mono Part or stereo/multichannel linked group at a time. It is used to change the duration of a Part (or stereo/multichannel linked group) without altering the pitch, within a range from 67% up to 150%. The length of the generated Take(s) can be specified in percentage, SMPTE, B/C/T or Tempo.

## Pitch Shift Process tool

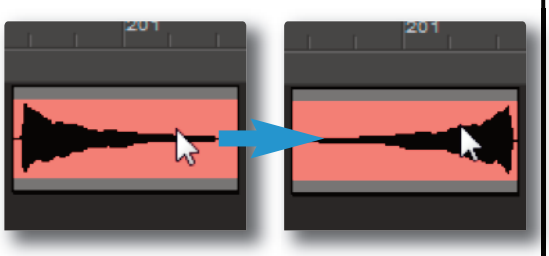
The Pitch Shift Process tool (***Ctrl+Shift+P***) only works on one mono Part or stereo/multichannel linked group at a time. It is used to change the pitch of a Part (or stereo/multichannel linked group) without altering its length, within a range from 67% up to 150%. The pitch of the generated Take(s) can be specified in percentage, or semitones and cents change.

### Sample Rate Convert Process tool

The Sample Rate Convert Process tool (***Ctrl+Shift+S***) only works on one mono Part or stereo/multichannel linked group at a time. It is used to resample a Part (or stereo/multichannel linked group), thereby changing length and pitch, within a range from 50% up to 200%. The length of the generated Take(s) can be specified in percentage, SMPTE, B/C/T or Sample Rate.

### Reverse tool

The Reverse tool (***Ctrl+Shift+R***) only works on one mono Part or stereo/multichannel linked group at a time. It is used to generate a new 'reversed' version of the selected Part (or stereo/multichannel linked group), along with the corresponding Take. It is similar to the process of playing the tape backwards. If the edited Part uses a Take that will not be referenced in the current Arrangement after the reverse process, you will be given the option to **Delete unreferenced Takes?**. Only click **Yes** if you are sure that the Take is not used in any other Arrangement.



### DC Removal tool

The DC Removal tool (***Ctrl+Shift+N***) only works on audio Parts. It is used to process a Part or a group of selected Parts using a high pass filter with a very low cut-off frequency. New Takes are generated which have all DC offsets removed (these may have been present when the Takes were originally recorded due to DC at mixer or preamp outputs).

### Mute tool

The Mute tool (***Ctrl+U***) works on audio and automation Parts. It allows any Part or group of selected Parts to be muted or de-muted by selecting the tool and clicking on that Part (or a Part in that group). Muted Parts retain the track assign data, and memorised track assignments are restored when the part is de-muted using the same tool. De-muting a Part may automatically mute active Parts that are wholly or partially in the same time range as that Part, depending on the current setting of the **Multiple Active Parts with same Output and Location** option (**Settings** / **Preferences** / **Arrangement**).

*NOTE: If you use the Mute tool on a group of Parts, the entire group will take on the same muted/active status as the Part that you click on, regardless of their prior status.*

### Glue tool

The Glue tool ***Ctrl+G*** works on audio and automation Parts. It allows you to remove cuts that have been made to a Part in the Arrange window. To glue a Part, assign the tool to a mouse button and click the Part section which is immediately before the cut. The cut will be removed and the glued Part will take on the parameter settings (ie., volume, fade settings, and track assignments) of the Part you selected.

It is only possible to glue together Parts that were originally edited using the cut tool, thus are from the same Take and are in the same order as the original Part (ie., continuous audio).

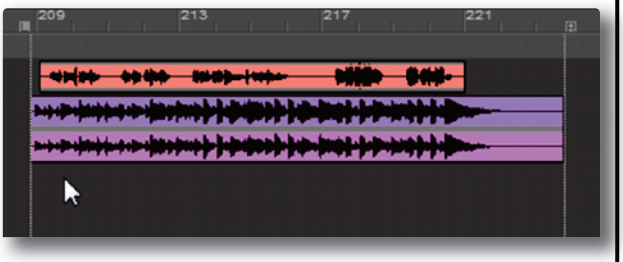
*NOTE: If you want to create one Part with one set of parameters, using Parts taken from different takes, then use the Consolidate tool to create a single new Take.*

## Mixdown tool

The Mixdown tool (***Ctrl+Shift+Y***) allows all active tracks to be digitally mixed together to create a new mono, stereo or multichannel Take, so that sub mixes or a complete stereo mix can be performed without leaving the digital domain. The Mixdown process does not introduce any extra background noise or degradation of the audio quality, and includes the effect of the volume, fade and mixer settings and automation associated with the original Parts. Mixdowns are performed in real-time, so that the result includes all the EQ, real-time FX (ie., plug-ins), volume and pan settings, automation data for the active Parts, and audio from external mixer inputs.

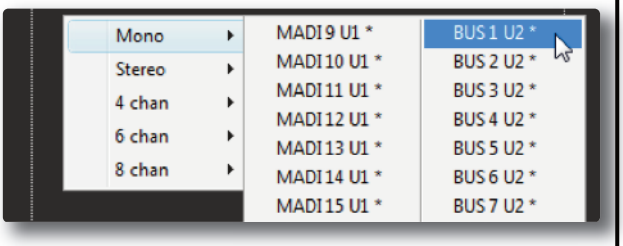
The Mixdown tool can only operate if enough physical tracks are available for the required destination format (mono, stereo or multichannel).

To Create a Mixdown, first position the Left and Right Locators at either end of the region you wish to mix. The length of the new Take is determined by the Left and Right Locators, so place them accurately to avoid using more disk space than you require.




Once you are happy with the volume, pan, fade settings and any plug-in settings and automation data, select the Mixdown tool and click between the Left and Right Locators, on the virtual track where you want the mixdown Parts to be positioned.

A menu will appear requiring you to select the mixdown format, and the output or bus to be used as the recording input. The output selection is critical, as only tracks routed to the selected outputs will be mixed down.




For example, you may have tracks 1 to 4 routed to busses 1-2, and tracks 5 to 8 routed to busses 3-4 in your Mixer. If so, selecting **Bus 1-2** will only mix down tracks 1 to 4.

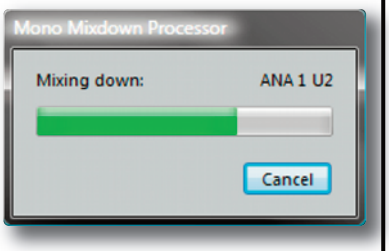
It is therefore a good idea to bus all of your tracks to a Master mixer column so they are all available from one output selection. You can then control which tracks will be mixed by selecting or muting them in the Arrange window.

*See the Mixer section of this manual for more information.*

*Remember, you will not get any sound without an active Mixer!*

Releasing the mouse button will initiate the mixdown. The lowest numbered track or track group which is not already assigned within the area defined by the Locators will automatically be selected for recording the mixdown to.

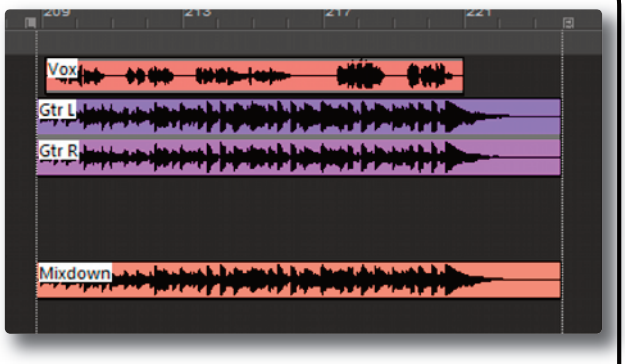
The dialogue box shown to the right will appear, showing the mix's progress. You will also hear the audio that has been selected.




The new Take(s) and Part(s) will be named **Mixdown**, with **L** and **R** extensions in the case of a stereo mixdown, along with a number if there are previous mixdowns in the selected folder.

The File Manager will also show that the Takes originated as a mixdown.

The Mixdown creates new audio data and saves it on disk as a Take, so it is not possible to extract the original tracks from the mix. However, the mixdown Part can be assigned to a physical track, and the original Parts can be left muted in the Arrange window, with the Takes still present on the hard disk. This means that you can always return to the original Arrangement and redo the Mixdown at any time.




*NOTES:*
*As the mixdown is real-time, you can include external signals that are fed into Soundscape's Mixer. These signals could be analogue Outboard, digitally connected FX processors, or another audio software synchronised via ASIO Positioning Protocol.*
*When using the Mixdown tool with the Automatic Delay Compensation (ADC) function active, the* ***Shift Audio Tracks if Automatic Delay Compensation is active*** *option is ignored. If ADC is not active, the Mixdown tool compensates for the audio path to the mixdown point that presents the longest delay. Please refer to the Automatic Delay Compensation section in this chapter for a detailed description of the ADC function.*

## Waveform tool

The Waveform tool (***Ctrl+W***) works only on audio Parts. It is used to turn on and off the waveform display for selected Parts in the Arrangement window, and to generate waveform display files (extension .wvf). These are stored in the C:\Soundscape\Wvf folder.

The waveform files are compressed for display by 512:1 compared to the sizes of Takes on disk, and therefore occupy a small amount of disk space on the PC. Once you have generated a waveform for a Part, it will be displayed in the Part.

As you zoom in, the waveform display is automatically switched from the low resolution PC files to the sample resolution waveforms stored on the disk. Fades and volume changes do not scale the waveform display, making it easy to view waveforms that have a low volume setting, or are at the end of a fade out.

You can select whether waveform files will be saved or discarded upon exiting the program with the option **Keep all generated waveform files** (**Settings** / **Preferences** / **General**).

## Noise Gate tool

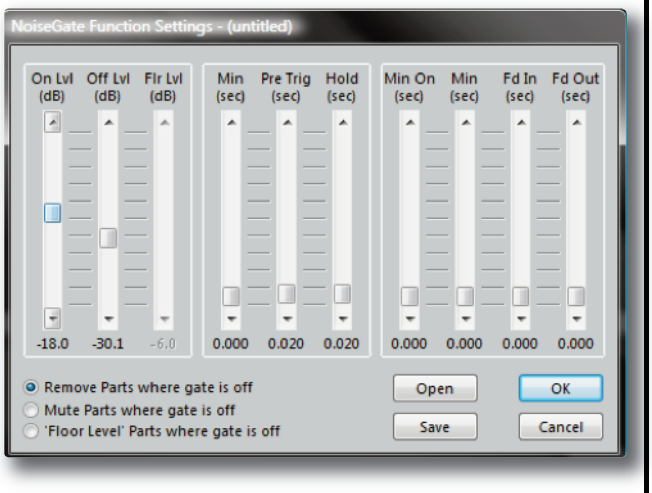
The post-processing Noise Gate function (***Ctrl+K***) works only on audio Parts. It is accessed when you click a Part or a group of Parts with the Noise Gate tool.

You can select a fader by either clicking on it or using the ***Tab*** or ***Shift+Tab*** key(s). The selected fader will flash and can be moved by either clicking and dragging it with the mouse, or using the arrow keys (for a slow/fine level change) or the ***Page Up***/***Page Down*** keys (for a fast/coarse level change). When using the arrow keys, ***Left Arrow*** and ***Right Arrow*** have the same function as ***Up Arrow*** and ***Down Arrow*** respectively.

The noise gate processes a Part or group of Parts, and automatically makes non-destructive edits to the Arrangement corresponding to the audio in the Part(s) and determined by the settings of the ten noise gate parameters. This generates multiple smaller Parts, but leaves the Takes on disk untouched. Standard gating techniques can be applied to the audio, but with some powerful additions such as programmable fades, pre-trigger time, and floor attenuation.

It is possible to store your gate settings for easy recall, using the **Open** and **Save** functions

**Noise Gate function parameters**




- **On Lvl** (–96..0dB): Threshold level for opening the gate.
- **Off Lvl** (–96..0dB): Threshold level for closing the gate (ie., the gate closes when the audio drops below this level).
- **Flr Lvl** (–96..0dB) (Floor Level) Level that is used for scaling Part volumes of Parts that are in the 'gate off' region. This setting is only used if the **"Floor Level" Parts where gate is off** option is selected.
- **Min Det** (0.000..10sec) (Min Detection Time): 'Gate on' regions less than this duration are ignored, so that fast glitches and clicks do not open the gate.
- **Pre Trig** (0.000..10sec) (Pre Triggering Time): Time that the gate opens before the **On Lvl** is reached (ie., allowing audio that starts just before the **On Lvl** is reached to be included).
- **Hold** (0.010..10sec): Time that gate remains open after the audio drops below **Off Lvl**. This is re-triggerable, so if the audio rises above the **On Lvl** within the hold time, the gate will remain open.
- **Min On** (0.000..10sec) (Minimum 'gate on' time): 'Gate on' region minimum length (ie., this sets the smallest Part length that will be generated).
- **Min Off** (0.000..10sec) (Minimum 'gate off' time): 'Gate off' region minimum length (ie., this sets the smallest gap or muted Part that will be generated).
- **Fd In** (0.000..10sec) (Fade in time): 'Fade in' Part length (can be shortened automatically if not enough space between Parts is available).
- **Fd Out** (0.000..10sec) (Fade out time): 'Fade out' Part length (can be shortened automatically if not enough space between Parts is available).
- **Remove Parts where gate is off**: All Parts that are in the 'gate off' region will be deleted.
- **Mute Parts where gate is off**: All Parts that are in the 'gate off' region will be muted.
- **"Floor Level" Parts where gate is off**: All Parts that are in the 'gate off' region will be scaled down in volume by the amount specified for the **Flr Lvl** parameter.

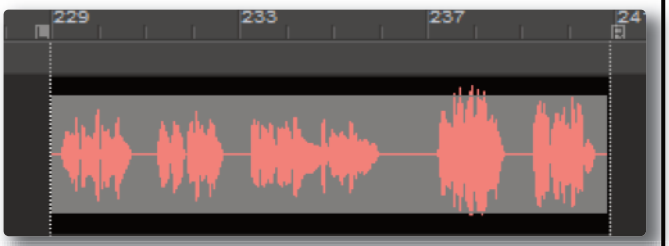
Adjust the parameters as required for the type of audio material you wish to process. To get an idea of the required threshold levels, open the Mixer window and look at the relevant peakmeter(s) while scrubbing or soloing the audio.

*NOTES:*
*You can use the Normalize function to check out the peak value of a selected Part, and then abort Normalizing by clicking* ***Cancel****. Remember, however, that the pre-normalizing peak detection process does not take Part volumes or fades into account.*
*The fades created by the Noise Gate tool use the fade curve defined with the Volume & Fade Edit tool. If required, the Volume & Fade Edit tool can be used after noise gate processing to select a different curve.*

A good starting point is to set the **On Lvl** to 15dB below peak level, and the **Off Lvl** to 15dB above the noise. The noise gate scans the audio in the Parts, taking into account the beginning volume of the Parts that exist before processing has taken place, ie., you can pre-adjust the volume, to vary the levels of any troublesome sections that do not gate effectively.

With inappropriate parameter settings, the Parts generated may not be what you intended. Remember that noise gate processing is non-destructive, so simply undo the edits, re-adjust the parameters and reprocess. It is always a good idea to make a copy of the original Part on another virtual track using the Copy Vertical tool before noise gate processing, just in case you perform a different operation afterwards and Undo (of the noise gate processing) becomes impossible.

In this example, a short vocal phrase was recorded:

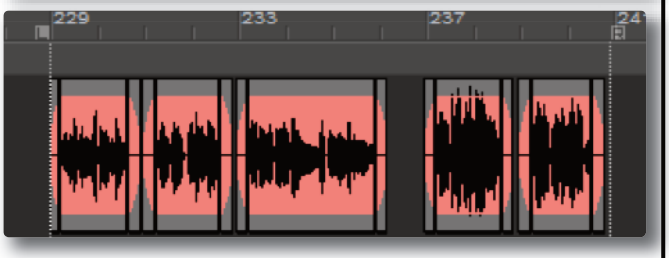



The noise gate processing was then finalized, but the chosen parameters produced too many short Parts, removing some of the wanted sections of audio:



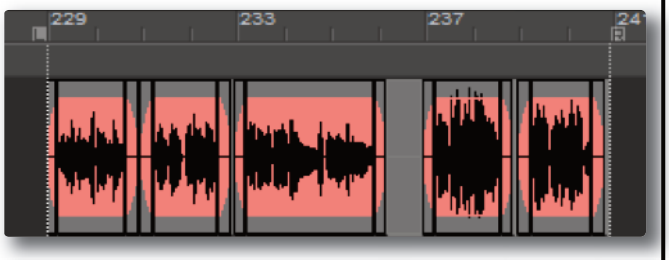

The noise gate parameters were modified to add a short fade in and fade out; The **Hold**, **Min On**, and **Min Off** times were increased and the thresholds slightly lowered.

Reprocessing achieved the desired result, so that a gap in the vocal was removed:

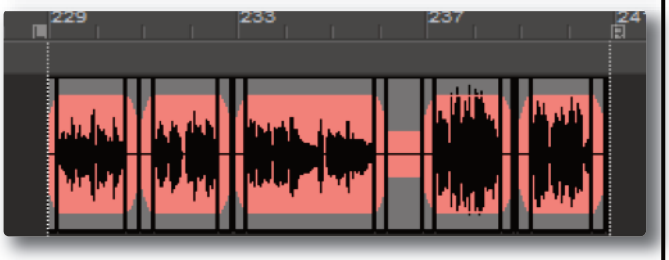



In this next example, **Remove Parts where gate is off** was selected.

If **Mute Parts where gate is off** is selected, then muted Parts are generated in the 'gate off' regions. This makes it easy to check the audio that is being removed, using the Solo tool.




If **Floor Level Parts where gate is off** is selected, then instead of being removed in the 'gate off' regions, the audio is just attenuated. For situations where completely removing the background ambience sounds unnatural, this can provide a controlled reduction of background levels.

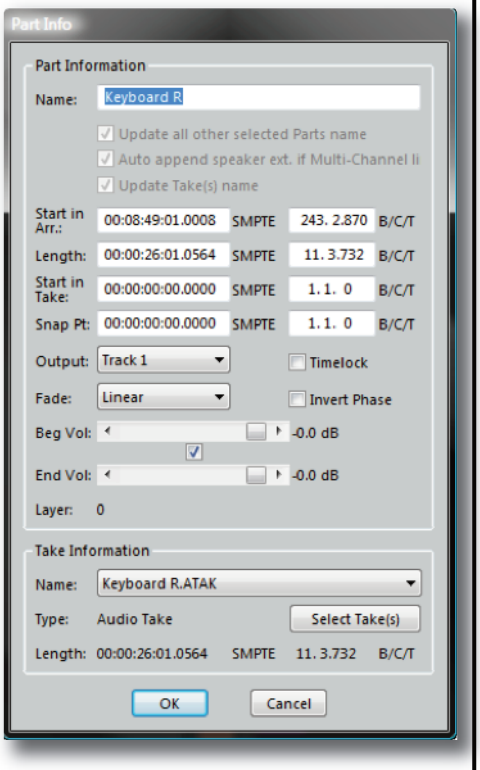


## Info tool

The Info tool (***Ctrl+I***) works on Audio and Automation Parts. It is used to view or edit information about a Part or Take. It allows naming of Parts and Takes, and precise editing of all Part parameters.

**Audio Parts**

To view Part information, select the tool and click on an individual Part. The **Part Info** dialogue box will be displayed:




Here you can enter a name for the Part and Take, move the Part to a new start time, edit the Part length, change output assignments, assign fade curves, adjust beginning and end volumes, invert the phase, set the timelock status, check the Part's layer assignment and select which audio Take the Part will use.

- If the **Update all other selected Parts name** box is checked, the edited name will replace the current name of all the selected Parts when **OK** is clicked.
- If the **Auto append speaker ext. if Multi-Channel linked** box is checked, an appropriate extension will be added to the name of all selected Stereo-Linked or Multi-Channel linked Parts. For Multi-Channel linked Parts, the extension may be 'numeric' or 'surround' (eg. L, C, R or S), depending on the setting of the **Multi-Channel speaker name extension** option under the **Settings** / **Preferences** / **General** menu.
- If the **Update Take(s) Name** box is checked, the name of the Takes referenced by the selected Part(s) will be updated when **OK** is clicked.

Clicking the **Select Take(s)** button will select the Take referenced by the Part in the File Manager. Please note that this may change the current folder selection. The **Select Take(s)** button is dimmed if several Parts are selected which reference Takes located in different folders (as they cannot be selected simultaneously).

The start position in the Arrangement, the length, and the start position in the Take can also be entered in bars/counts/ticks, or in SMPTE + sample extension, allowing sample accurate positioning.

If multiple Parts are selected, the **Part Info** window shows the values for the Part that was clicked. The amount of change entered for any parameter is applied for all other selected Parts where possible. For example, if the Part length is reduced by 1 frame, all other selected Parts also have their length reduced by 1 frame. The only exceptions to this are when the same amount of change to the **Start in Arr**, **Length**, or **Start in Take** values cannot be made for other selected Parts (because of overlaps, Part/Take lengths, etc.). The maximum amount of change that can successfully be made to all selected Parts is substituted. A way of checking that the values entered are applicable to all Parts is to click on another parameter after entering changes – If the changed value remains as you have entered it, then the change is possible for all selected Parts; If not, then the value represents the maximum allowable change.

Changes to the Take assignment are not allowed when multiple Parts are selected, including stereo linked Parts. Changes to the **Output**, **Timelock**, **Fade** and **Invert Phase** settings are only applied to the Part that was clicked.

**Automation Parts**

The Info tool works in a similar manner on Automation parts, though the available parameters are different, according to the nature of the Part. The **Fade**, **Volume** and **Phase** parameters are absent, while a **Display** parameter is added, which determines what type of automation data is shown in the Part (the Automation Curve Select tool, described in the '*Mixer Automation*' section of this manual, can also be used for that purpose).

## Consolidate Process tool

The Consolidate Process tool (***Ctrl+Shift+G***) only works on audio Parts. It is used to create new Takes by combining the audio referenced by selected Parts that are located on the same virtual track and assigned to the same physical track. One new Take is created for each physical track.

The audio data included in the new Takes reflects the Part volume information, fades and crossfades that were applied to the selected Parts. Any silent sections between the selected Parts are also included in the new Takes.

After the new Takes have been generated, the selected Parts in the Arrangement are edited in such a way that they reference the new Takes, and so that overlaps, fades, crossfades and silences are removed, always resulting in the smallest possible number of Parts.

Optionally, the process also offers an option to delete selected muted Parts when present.

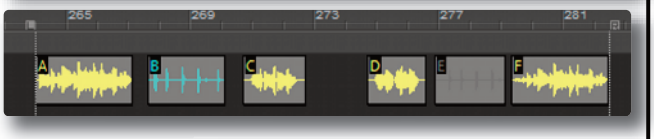
Any processed Parts which become muted are automatically deleted.

> ***WARNING:*** *The Takes generated by the Consolidation process only include the audio data that is actually used by the selected Parts. If the original Takes referenced by these Parts contain valuable audio data that is not used by the Parts in the current Arrangement, please note that this audio may be deleted after processing, depending on the option selected under* ***Settings*** */* ***Preferences*** */* ***Arrangement*** */* ***Audio Processing tools behaviour*** */* ***Audio Takes that become unused after processing****.*

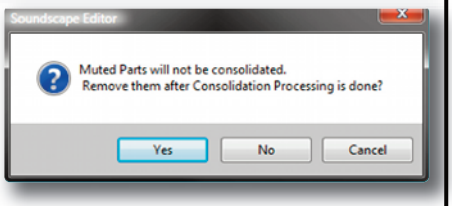
In this example, Parts A, C, D and F are assigned to track 9. Part E is also assigned to track 9 but is muted. Part B is assigned to track 16.




Parts A, B, C, D, E and F are selected for consolidation:




Because the selection includes a muted Part, the dialogue box shown right is displayed after clicking one of the selected Parts. Clicking **Yes** or **No** triggers the processing; Clicking **Cancel** aborts the operation.



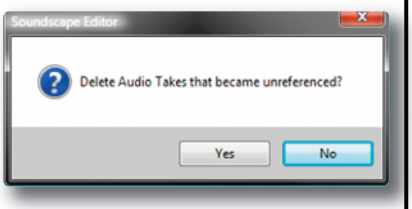

Clicking **Yes** allows Part E to be removed from the Arrangement, resulting in only three parts being selected. Parts C, D and F are combined to create an new Take which is referenced by Parts A #01 and C #01. Part B is also replaced by a new Part B #01, which references a new Take.




Clicking **No** allows the muted Part E to remain in the Arrangement.



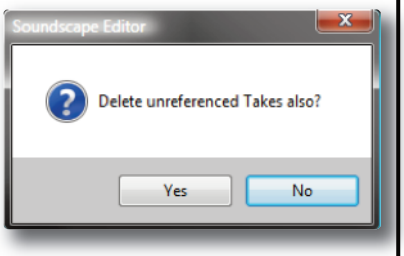

A further dialogue box may be displayed, depending on the option selected under **Settings** / **Preferences** / **Arrangement** / **Audio Processing tools behaviour** / **Audio Takes that become unused after processing**. The available options are described in the *Settings Menu* chapter.



## Delete tool

The Delete tool (***Ctrl+D***) works on Audio and Automation Parts.

Click on a Part or group of selected Parts to delete it. If Takes are left on the record/process folder which have become unreferenced in the current Arrangement, a dialogue box will be displayed (shown right).




If you no longer require these Takes, and you are sure that they are not used by any other Arrangement, click **Yes** and the Takes will be deleted from disk. Click **No** if you want to save them, or if you are not sure whether they are used elsewhere.

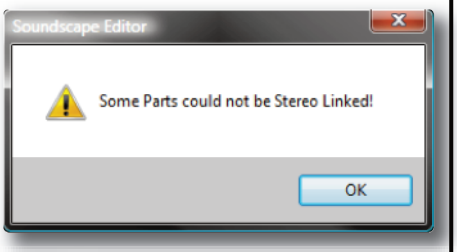
*NOTE: When a Part or group of Parts has been selected, you can simply use the **Delete** key instead of the Delete tool.*

## Stereo Link tool

The Stereo Link tool (***Ctrl+L***) only works on audio Parts. It allows two Parts to be linked for all editing operations, including soloing.

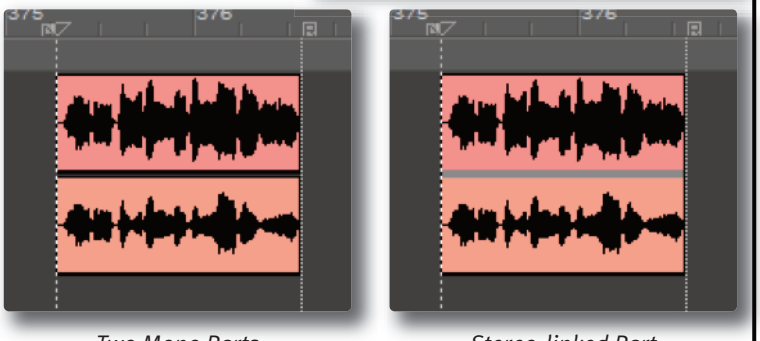
To stereo-link two Parts, select them then click either of them. If the target Parts are surrounded by other Parts of identical length and start position on adjacent virtual tracks, click on the uppermost targeted Part.

Stereo linking can only take place if the two Parts are on adjacent virtual tracks, have identical lengths and start positions in the Arrangement, and reside on the same Layer. If this is not the case, a dialogue box appears warning that the linking could not be completed.




Multiple Parts may be selected, but only those fulfilling the above criteria will be stereo linked. Linked Parts appear as shown to the right, still with the original track assignments and colours. Using the Track Assign tool on stereo linked Parts will assign tracks in odd-even pairs (ie., 1-2, 3-4, etc.).

The same tool can be used to undo the stereo link between two Parts – Just click either of the linked Parts and they become independent again.

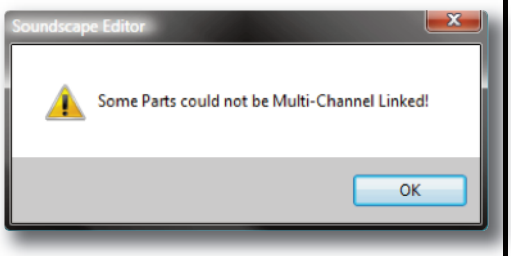



*Two Mono Parts* *Stereo-linked Part*

*NOTE: If you use the Stereo Link tool on a group of Parts, the entire group will take on the same link status as the Part that you click on, regardless of each Part's prior status.*

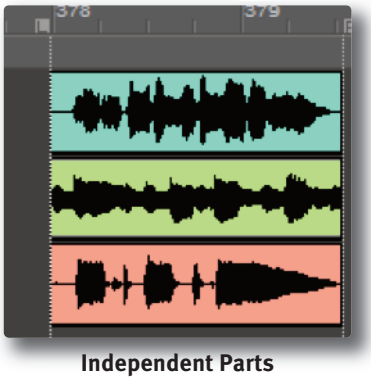
## Multi-Channel Link Part tool

The Multi-Channel Link Part tool (***Ctrl+4***) only works on audio Parts. It allows multiple Parts to be linked for all editing operations, including soloing. To multi-channel link Parts, select them then click one of them. Multi-channel linking can only take place if the Parts are on adjacent virtual tracks, have identical lengths and start positions in the Arrangement, and reside on the same Layer. If this is not the case, a dialogue box appears, warning that the linking cannot be completed.

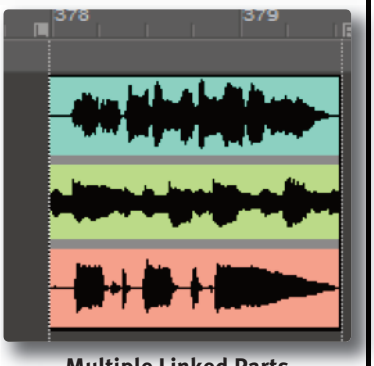



Multiple Parts may be selected, but only those fulfilling the above criteria will be multi-channel linked. Any linked Parts appear as shown on the right side, still with the original track assignments and colours. Using the Track Assign tool on multi-channel linked Parts will assign tracks in series (ie., 1-2-3, 3-4-5, etc.).

The same tool can be used to undo the multi-channel link between Parts – Just click either of the linked Parts and they become independent again.



**Independent Parts**



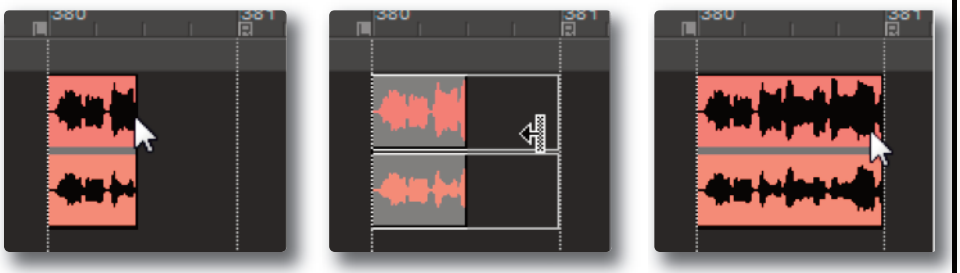
**Multiple Linked Parts**

*NOTE: If you use the Multi-Channel Link tool on a group of Parts, the entire group will take on the same link status as the Part that you click on, regardless of each Part's prior status.*

## Trim tool (exclude Adjacent Parts unless selected)

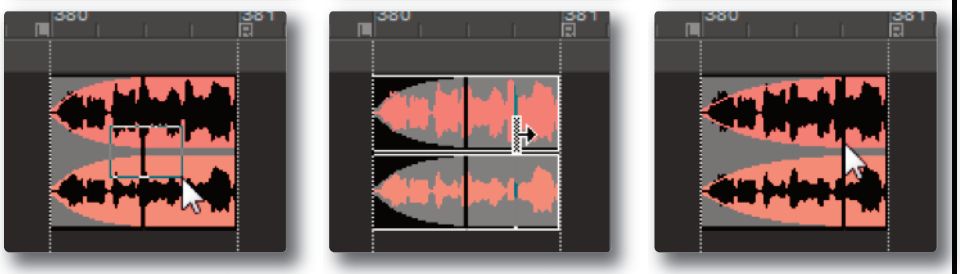
This tool, also accessed using the ***Ctrl+T*** keys, works on Audio and Automation Parts. It allows the start or end of a Part or group of selected Parts to be dragged to the left or right using the mouse, thus shortening or lengthening the Part(s). The audio or automation data remains in the same timing position, allowing you to 'reveal' more or less of the data from the Take.

To trim a Part, click on its beginning or end, hold the mouse button down and drag with the mouse in the required direction.




You can also move the position of a cut between two Parts (eg., when you have cut a Part in two with a fade into the second section).

The tool always allows you to draw a selection box across a cut, to select the Parts on both sides.




Dragging the cut will lengthen one Part and shorten the other.

You can trim Parts from multiple tracks at the same time, as long as the beginnings/ends/cuts are at identical time positions.

### Trim tool (include Adjacent Parts)

This tool, also accessed using the ***Ctrl+Alt+T*** keys, works on Audio and Automation Parts. It works exactly as the Trim (Exclude Adjacent Parts unless selected) tool described above, except that when a Part is trimmed, any adjacent Part on the same virtual track is trimmed at the same time, whether or not it has also been selected.

*NOTE: If the previous or next Part on the same virtual track is not adjacent to the edited Part, ie., if there is a gap between the edited Part and the other Part, the Trim (Include adjacent Parts) tool will stop at the boundary of the other Part.*

### Using the Trim tools

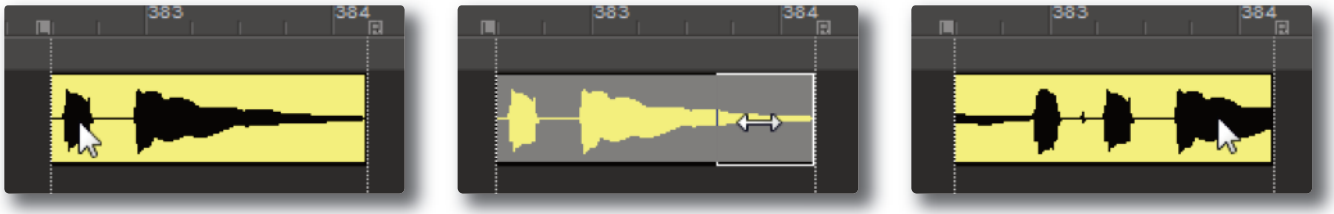
If Snap is active, a Part or a group of selected Parts can be trimmed in increments defined by the current snap value by clicking and holding it with the tool and using the ***Left Arrow*** and ***Right Arrow*** keys. If Snap isn't active, increments of one sample will be used.

The length of a Part can only be extended if it is shorter than the referenced Take, ie., if there is still available audio data on the disk.

*NOTES:*
*The Trim tools can also be used to adjust the length of a crossfade.*
*If a Trim tool seems to have no effect, it may be because you have clicked too far from the beginning or end boundary of the Part.*

### Slip tool

The Slip tool, also accessed using the ***Ctrl+J*** keys, works on Audio and Automation Parts. It allows the audio or automation data within a Part or group of selected Parts to be moved without changing the length of the Part or its start position. It is only possible to slip the data to the left or right if the Take on disk is longer than the Part in the Arrangement, and there is therefore excess data on at least one end.




Select the Slip tool, click and hold on the audio (waveform) that you wish to move, and drag left or right. The pointer shape will change to a double-pointed arrow, and an outline will be drawn, showing how much movement is being made (as shown in the second picture above). If Snap is active, its current setting is used, but applies from the start or end of the Part (depending upon the slip direction), rather than being relative to the Time Axis.

If Snap is active, the data within a Part or group of selected Parts can be slipped in increments defined by the current snap value by clicking and holding the Part (or one Part in the group) with the tool and using the ***Left Arrow*** and ***Right Arrow*** keys. If Snap is inactive, increments of one sample will be used.

*NOTE: Once the data within a Part has been moved using the Slip tool, the Part cannot be glued back to any previous/next Part, even if they were originally the same Part, because a discontinuity has been introduced in the audio where the Parts would join. If you need the data from two such Parts to be included in one single new Part, use the Consolidate tool.*

## Phase Invert tool

The Phase Invert tool (***Ctrl+P***) only works on audio Parts. It allows individual Parts to be inverted in real-time (ie., this is a playback parameter and there is no need for a new Take to be generated on the disk). This may be necessary if the signal is being inverted elsewhere in the record or playback audio path.



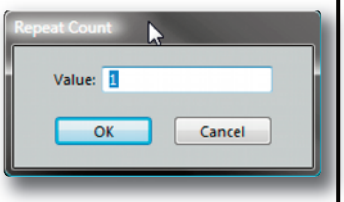

The Phase Invert status of a Part is reported in the **Part Info** window, where it can also be changed.

*NOTES:*
*If you use the Phase Invert tool on a group of Parts, the entire group will take the same Phase Invert status as the Part that you click on, regardless of their prior status.*
*Phase inversion can also add some interesting effects if a Part is copied, assigned to a new track, phase inverted and then is delayed slightly and played at the same time as the original. This introduces a simple comb filter which cancels some frequencies and enhances others, depending upon the delay.*

## Repeat tool

The Repeat tool (***Ctrl+R***) works on Audio and Automation Parts. It allows a Part or a group of selected Parts to be repeated any number of times in the Arrangement.

If Snap is active and set to musical or SMPTE time divisions, the repeats will start on the corresponding snap points after the end of the original Part. If Snap isn't active, (or if the selected snap value is shorter than the edited Part or group of Parts), the repeats will be adjacent, with no gap between the original Part and the succession of repeats. Clicking the Part to be repeated calls up a dialogue box (see right), where the required number of repeats must be entered before clicking **OK**.




*NOTE: If the number of repeats entered causes an overlap with other Parts in the Arrangement and overlapping Parts are not allowed under **Settings** / **Preferences** / **Arrangement** / **Overlapping Parts (for new edits)**, a message will be displayed warning that the maximum number of repeats possible without overlapping will be made.*

## Timelock tool

The Timelock tool (***Ctrl+Shift+L***) works on Audio and Automation Parts. It is a security feature that allows a Part or group of selected Parts to be locked in time in the Arrangement, so that they cannot be accidentally moved, slipped or deleted. Other edits can be performed as normal on these Parts.

A timelocked Part has a star (*****) character appended to its name as shown on the right screenshot.

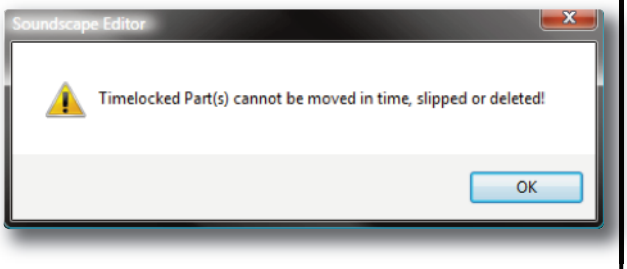



*NOTE: If you use the Timelock tool on a group of Parts, the entire group will take on the same timelocked status as the Part that you click on, regardless of their prior status.*

## Recording with Timelocked Parts in the Arrangement

Timelocked Parts respond as normal to the Track Assign, Volume Trim, Fade In/Out, Copy, Trim, Mute, Move Vertical tools. However, any edit that would cause the data in the Part to be changed in timing or would remove the Part from the Arrangement is impossible.

For instance, if you start a recording with manual punch out that potentially could overwrite a timelocked Part in the Arrangement, the message shown right will be displayed.




Recording will only be allowed if you select Auto Punch Out and the Right Locator is before the timelocked Part. Alternatively, move the record track(s) to different virtual tracks which do not contain Timelocked Parts.



### Automation Curve Select tool

This tool, also accessed via the ***Ctrl+Shift+A*** keys, only works on Automation Parts. It is used to select which automation curve is displayed for an automation Part which contains several automation curves.

*This tool is described in detail in the 'Mixer Automation' section of this manual.*



### Automation Event Editing tool

This tool, also accessed via the ***Ctrl+Shift+E*** keys, only works on Automation Parts. It is used to create, delete or edit automation events within existing automation Parts.

*This tool is described in detail in the 'Mixer Automation' section of this manual.*



### Automation Events Thinning tool

This tool, also accessed via the ***Ctrl+Shift+F*** keys, only works on Automation Parts. It is used to thin automation data by a user-selectable factor.

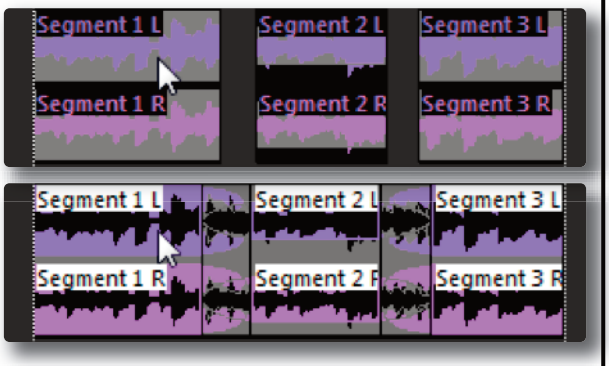
*This tool is described in detail in the 'Mixer Automation' section of this manual.*

## Crossfade tool

The Crossfade tool (***Ctrl+Shift+X***) only works on audio Parts. It allows the crossfading of any number of selected successive audio Parts on the same virtual track, providing these Parts overlap. If the Parts do not overlap but reference Takes that have enough excess audio at their respective end of the projected crossfade area, they will be trimmed automatically when the Crossfade tool is used, so that the necessary overlap is created. Therefore, the Parts do not need to be adjacent before creating the crossfade. Note that it is even possible for a mono Part to be crossfaded with a stereo-linked Part.

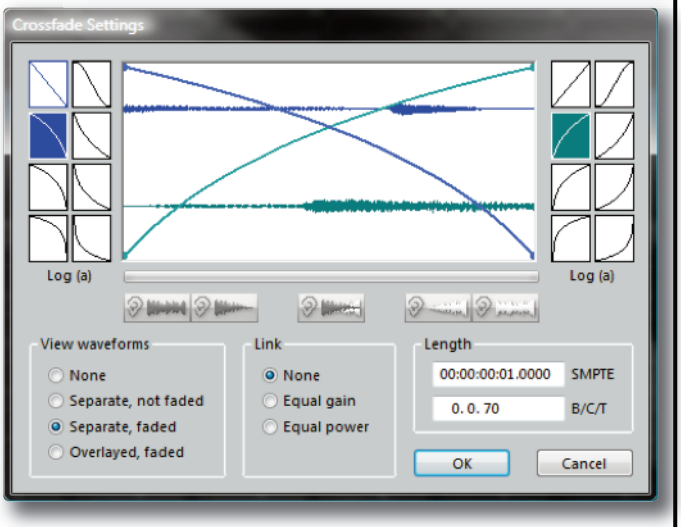
Crossfaded Parts can be assigned to the same physical track. This means that the number of available physical tracks is preserved during a crossfade (the number of Parts being played back momentarily exceeds the number of playback tracks).

In the following example, three consecutive pairs of stereo-linked Parts with no overlaps are selected, then the Crossfade tool is used.




The necessary overlaps are created automatically because the Parts reference Takes which contain enough audio for the automatic trimming process to be performed. Notice how the volume contour of the **Segment 2** Parts is preserved, as shown by the amplitude of the coloured section of the Parts.

Double-clicking on the Crossfade tool icon will call up the **Crossfade Settings** window, where you can define the default settings that are applied to each crossfade you subsequently create. A similar window can be accessed by clicking with the tool in any crossfade in the Arrangement, so that each crossfade can have its own settings.




The top section of the window shows a graphical representation of the crossfade. The fade curve and waveform for the Fade Out and Fade In Parts are shown in blue and green respectively. The eight possible curve shapes are shown on either side of the graphic. The currently selected curve is indicated by a blue (Fade Out) or green (Fade In) background. If **Equal power** is selected in the **Link** section of the window, unavailable curve shapes will appear dimmed. If **Link** is set to **None**, the start and end points of each fade curve are independently editable in the window by clicking and dragging them with the mouse (so that the fade out and fade in can have a different length and position from one another within the crossfade itself).

*NOTE: Any existing volume settings for the crossfaded Parts are preserved up to the crossfade start point and from the crossfade end point. For example, if the Fade Out Part has a volume of -4dB at the crossfade start point, the fade out will be from -4dB to -96dB.*

The five playback buttons in the middle of the window allow you to hear:

- The fade out on its own, fade NOT applied.
- The fade out on its own, fade applied.
- Both Parts with the crossfade applied.
- The fade in on its own, fade applied.
- The fade in on its own, fade NOT applied.

*NOTE: If you have accessed the **Crossfade Settings** window by double-clicking on the Crossfade tool, the playback buttons are unavailable (dimmed) because you are adjusting default settings and there is no corresponding audio to be played back.*

### View waveforms

In this section of the window, you can choose preferences for when and how the audio in the crossfaded Parts is displayed. Note that the terms 'faded' and 'not faded' only describe how the audio is shown on screen and does not affect playback.

### Link

- If you select **None**, any fade curve shape can be chosen independently for the fade out and fade in.
- If you select **Equal gain**, you can choose any fade curve shape for either the fade out and fade in, and the fade curve shape for the other one will be selected automatically so that it retains equal overall gain. For example, if the fade out's gain drops slowly, the 'fade in's gain will increase at the same slow rate.
- If you select **Equal power**, you can choose one of three possible fade curve shapes (unavailable ones will appear dimmed) for either the fade out or fade in, and the fade curve shape for the other one will be selected automatically so that the perceived volume does not vary during the crossfade. However this has practical limitations: while it is easy to maintain a constant perceived volume if the audio in both Parts consists of test tones of equal volume in the first place, a crossfade algorithm that can keep the power constant in all cases simply does not exist. An extreme example would be a crossfade between two sine waves of the same frequency, where one is phase shifted by 180 degrees. The power cannot be constant in this case, no matter what fade curve shapes are applied. The **Equal power** option has been implemented primarily in the interest of compatibility with other systems.

### Length

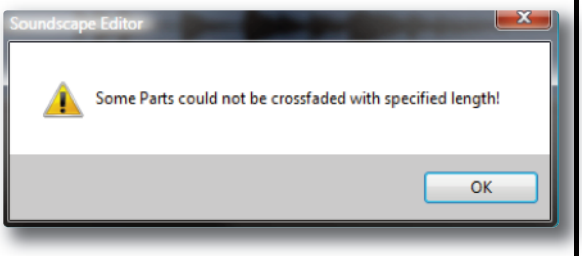
When a crossfade is created between non overlapping Parts, its length can be specified before using the tool, in the **Crossfade Settings** window accessed by double-clicking on the Crossfade tool icon. The required crossfade length can be entered either in SMPTE + sample extension or in B/C/T (bars/counts/ticks) and will be applied subsequently for each use of the tool.

When a crossfade is created between overlapping Parts, the start and end points of the overlap become the start and end points of the crossfade. The length setting as defined in the **Crossfade Settings** window for the tool is ignored.

*NOTE: In the case of a new crossfade, If the Parts or the referenced Takes are too short to produce the required crossfade length, the maximum possible length will be generated instead.*

For any existing crossfade, the length can be altered in the **Crossfade Settings** window accessed by clicking with the tool on that particular crossfade. Whether you increase or reduce the length of a crossfade, the start and end points are moved by the same amount in the relevant direction to reach their new positions, while the center point of the crossfade remains fixed. If the crossfaded Parts are too short to reach the required length but reference Takes that have enough excess audio at their respective end of the crossfade, the Parts will be trimmed automatically so that the required length can be obtained.

For an existing crossfade, if you enter a value that cannot be reached because the Parts and their referenced Takes are too short, this warning message will appear.




*NOTE: The Trim tools can be used to adjust the length of a crossfade.*

## Context Sensitive Edit tool

The Context Sensitive tool (***Ctrl+2***) works on Audio and Automation Parts. By default, this single, fully customisable time-saving tool combines the functions of the 'Volume Trim', 'Slip', 'Move', 'Solo', 'Trim (Include Adjacent Parts)', and 'Automation Curve Select' tools, switching between these functions automatically, according to its current position on the Part to be edited and that Part's type (audio or automation). If the ***Alt*** key is held down while moving the pointer, the functions of the 'Mute', 'Track assign', 'Copy', 'Cut', and 'Trim' tools are available instead. The pointer shape changes as it is moved over the Part to indicate in advance which type of edit can be performed. The pointer shapes clearly indicate the function, and are often identical to the ones shown when using the dedicated tools. In addition, zooming and scrolling can always be performed with the same tool and menus which provide access to almost all of the remaining editing or processing functions can be called up using the right mouse button.

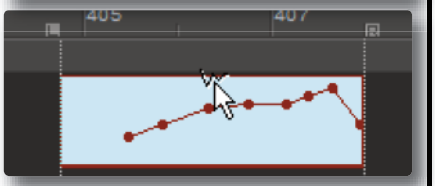
*NOTE: Selecting the Context Sensitive Edit tool will automatically cancel all previous tool selections. This is normal and due to the fact that the Context Sensitive Edit tool makes use of both mouse buttons and the **Alt** key.*

### Using the left mouse button

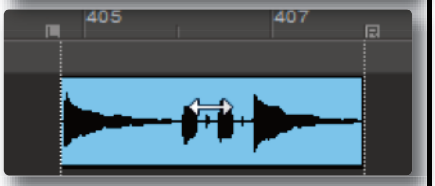
If the pointer is in the upper section of an audio Part, the Context Sensitive Edit tool operates as the Volume Trim tool.



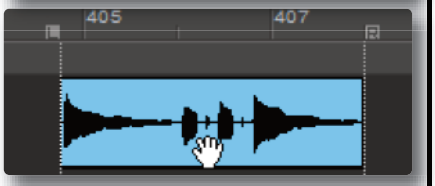

If the pointer is in the upper section of an automation Part, the Context Sensitive Edit tool operates as the Automation Curve Select tool.



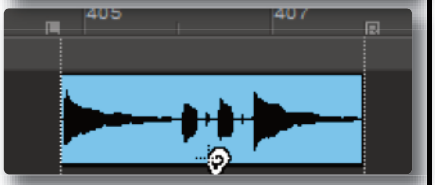

If the pointer is in the upper middle section of a Part, the Context Sensitive Edit tool operates as the Slip tool.



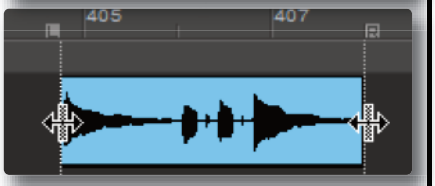

If the pointer is in the lower middle section of a Part, the Context Sensitive Edit tool operates as the Move tool:



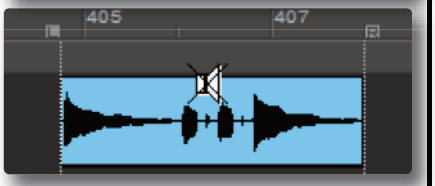

If the pointer is in the lower section of a Part, the Context Sensitive Edit tool operates as the Solo tool:



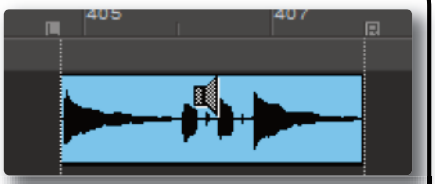

If the pointer is near the beginning or end boundary of a Part, the Context Sensitive Edit tool operates as the Trim (Include Adjacent Parts) tool:



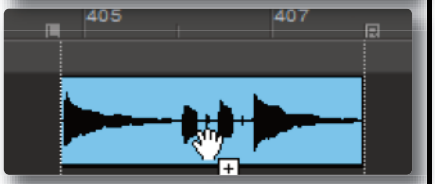

With the ***Alt*** key held down, if the pointer is in the upper section of a Part, the Context Sensitive Edit tool operates as the Mute tool:



With the ***Alt*** key held down, if the pointer is in the upper middle section of a Part, the Context Sensitive Edit tool operates as the Track Assign tool:



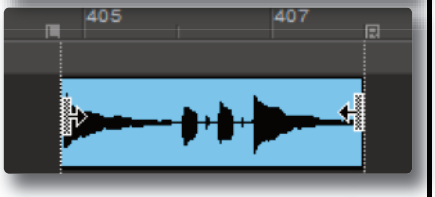

With the ***Alt*** key held down, if the pointer is in the lower middle section of a Part, the Context Sensitive Edit tool operates as the Copy Tool:




With the ***Alt*** key held down, if the pointer is in the lower section of a Part, the Context Sensitive Edit tool operates as the Cut tool:

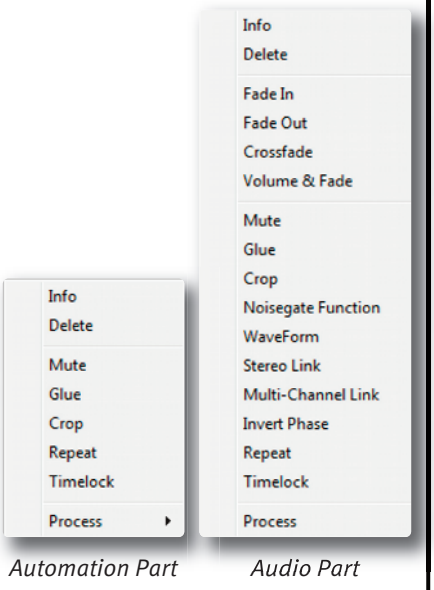



With the ***Alt*** key held down, if the pointer is on the edge at the beginning or end of a Part, the Context Sensitive Edit tool operates as the Trim tool:



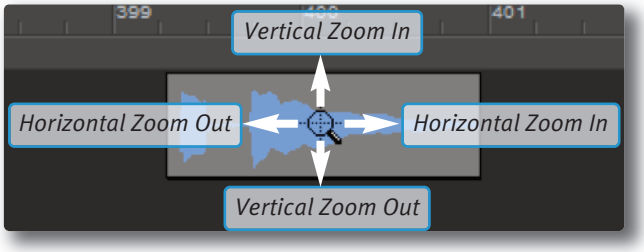

**Using the right mouse button**

Right-clicking on a Part causes a context sensitive menu to appear, which allows for nearly all other possible editing or processing operations to be carried out. This menu is different according to the type of track being edited (audio or automation):

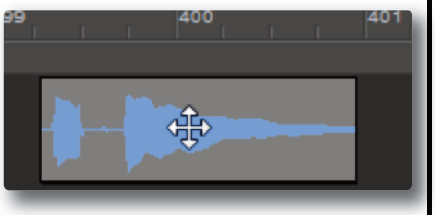



*Automation Part* *Audio Part*

Dragging a Part with the right mouse button adjusts the horizontal or vertical zoom level according to the dragging direction. The point where the Current Locator is located when the dragging movement starts is the 'center' of the zoom (ie., it does not move and therefore remains visible). If you are aiming for a particular edit point within a Part, clicking and dragging that point will keep it in focus:

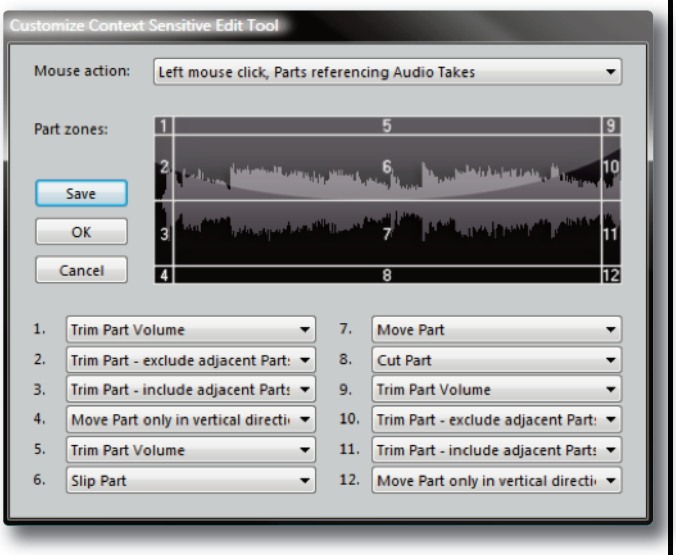



With the ***Alt*** key held down, dragging a Part with the right mouse button causes the Arrange window to scroll horizontally or vertically in relation to the dragging direction:



### Customising the Context Sensitive Edit tool

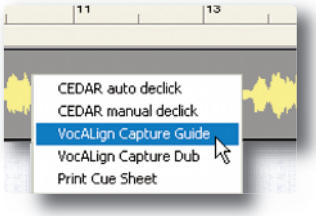
Once you have got to grips with the Context Sensitive Edit tool, you may want to customise its operation to suit your preferences. This can be done easily: double-clicking the Context Sensitive Edit tool icon in the Toolbar opens a window where you can select any editing function for each one of twelve predefined Part zones, and for each one of four possible left mouse button actions (with or without using the ***Alt*** key, for audio Parts or automation Parts). Your customisation choices will be saved to the .INI file and become permanent if you click **Save**, while clicking **OK** only keeps them active for the current session.




## XPro tool - External off-line processing

This tool, also accessed using the ***Ctr+Alt+E*** keys, only works on audio Parts. It provides a software interface specification that allows third party companies to produce tools and software plug-ins for the SSL Soundscape.

There are several optional plug-ins that use the XPro interface, for example Vocalign by Synchro Arts, RenoVator from Algorithmix and Declick by Cedar.

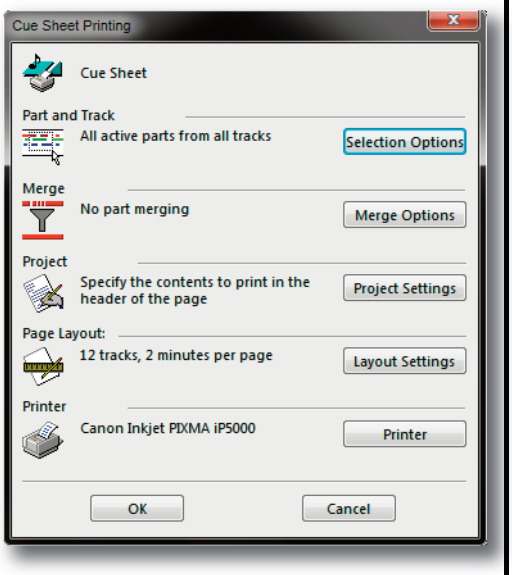



When installed, these plug-ins appear in a pop-up menu when you click on a Part or group of Parts with the XPro tool, along with the XPro Cue Sheet Printing plug-in which is included as standard and installed with Soundscape.

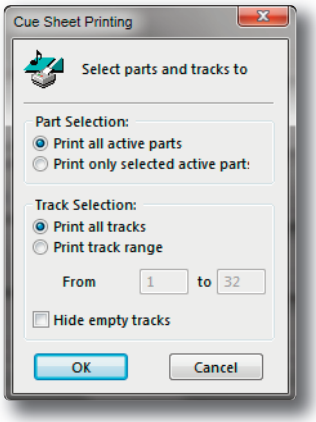
The third Party XPro Plug-ins are optional tools and they appear in the Optional Modules window accessible from the Options menu. Normally, you have to purchase a password in order to use a Plug-in, an exception is VocAlign, where Synchro Arts decided to use a challenge/response type authorisation procedure.

## XPro Cue Sheet Printing

Clicking on a Part in the Arrange window with the XPro tool calls up the XPro Tools menu, shown above. Selecting **Print Cue Sheet** calls up the **Cue Sheet Printing** window, shown right.

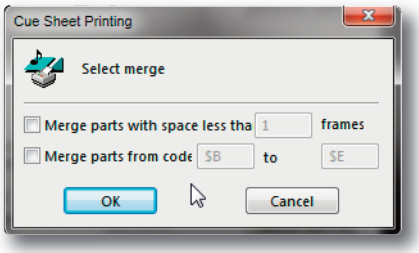



Clicking each one of the five buttons on the right of the window opens a further window in which parameters can be set for printing a cue sheet for the current Arrangement, as described below and over the page.




Clicking **Selection Options** opens the window shown left, which can be used to specify which active Parts and tracks are going to be featured on the printed cue sheet. Empty tracks can be hidden.

Click **OK** to save your settings, or **Cancel** to close the window without saving.

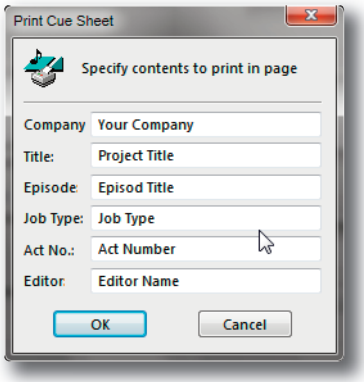



Clicking **Merge Options** opens the window shown left.

If the **Merge parts with space less than**... box is checked, Parts that are separated by less than the number of frames specified will be merged when printing.

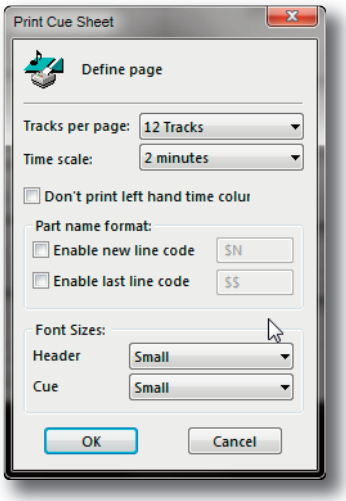
If the **Merge Parts from code**... box is checked, Parts between and including the ones that have the chosen codes specified in their name will be merged when printing.

Click **OK** to save your settings, or **Cancel** to close the window without saving.




Clicking **Project Settings** opens the window shown left, which can be used to enter relevant information about your company, the project and the editor.

Click **OK** to save your settings, or **Cancel** to close the window without saving.




Clicking **Layout Settings** opens the window shown left. In the top section of this window, the number of tracks and the time scale per page can be specified. Options are:

– 4, 6, 8, 10, 12, 16, 20, 24, or 32 tracks.
– 30 sec, 1 min, 2 min, 5 min, non linear.

It is possible not to print the left hand time column in order to save space.

If the **Enable new line code** box is checked, the specified code entered in a Part name forces a new line when printing.

If the **Enable last line code** box is checked, the specified code entered in a Part name causes the last part of the Part name to be printed at the bottom of Part.

In the **Font sizes** area of the window, font sizes for header and cue can be specified.

Click **OK** to save your settings, or **Cancel** to close the window without saving.

Clicking **Printer** opens a standard Windows Page Setup dialogue, which can be used to specify a paper source and size, select the orientation, and set margins. Click **OK** to save your settings, or **Cancel** to close the window without saving.

When all the parameters are set, clicking **OK** in the **Cue Sheet Printing** window will trigger printing. Click **Cancel** to close the window without printing or saving any of the settings.

# Mix Window

Soundscape's Mixer runs entirely on the DSP chips of the MX4 or Mixpander cards. The Mixer structure and routing are user definable and offer unrivalled flexibility for the use of real-time plug-in effects, EQs and processors. VST plug-ins running on the host CPU can also be inserted directly into Soundscape's Mixer environment.

As MadiXtreme cards have no DSP, their mixer functionality is significantly simplified. Their routing is 'hardwired' so that the MADI input and output is always the same number as the track. In addition, no DSP plug-ins are available.

It is easy to create and save Mixers, and the entire configuration with all current parameter settings can be saved at any time for 'total recall'. Unlike any other DAW, the Mixers are stored separately from the Project, making it possible to load different Mixes for a Project. Also unlike other DSP based DAW's, it takes less than a second to load a complete Mixer, including all Plug-Ins, complex Bus Routing etc.

The Mixer also supports the full range of automation created inside the Arrange Window, including all Controls and Buttons of inserted VST Plug-Ins. Each mixer strip can have any number of real-time processes running (eg. EQ, compressor, reverb, etc.) limited only by the available DSP power. For example, up to 128 mixer strips (we call them Mixer Columns) and 64 internal buses can be configured to send signals to internal real-time effects, with 99 selectable fader groups and solo groups.

You can also use any MX4 outputs and inputs as sends/returns to and from external effects units, while mixing tracks from your Soundscape Arrangement.

The Mixer inside Soundscape is also capable of working as a Mixing front end for any other ASIO DAW Software.

## Creating, Loading and Saving Mixes

By default, the **Soundscape def.mix** Mixer is loaded on the first startup. **Do not overwrite this Mixer**, as it is a good starting point to create your own Mixers.

The last saved Mixer (along with the last saved Arrangement) will be loaded on next startup of Soundscape.

**Soundscape def.mix** features 8 stereo and 8 Mono mixer strips routed to a 'Master' mixer strip, ready to mix 24 tracks from the Windows application of your choice down to outputs 1/2 of your MX4 or Mixpander (Unit 1, if several cards are present). 5 Stereo Send Buses and 3 Stereo Artist Monitors complete the Mixer and allow overdubs with near-zero latency Monitoring and global EFX.

You can load different mixers from the Mixers/Soundscape directory to experiment with, by clicking **Open Mix** under the **File** menu (or double clicking the Mix Files inside the File Manager window).

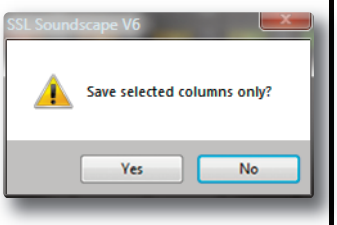
The default path for Soundscape mixer files is: **C:\Soundscape\Mix\Soundscape**

In order to create a new, blank Mixer, click **File** / **New Mix...**

In order to save your changes, click menu **File** / **Save Mix as**...

### Saving Selected Columns

You can also save a single column, or selected columns, as an independent mix file. To do this, hold ***Shift*** and select the column (or group of columns) you wish to save, then select **File** / **Save Mix as..** A pop-up will appear, asking if you want to save just the selected columns. Press **Yes** to continue, and name the file in the standard 'Save As' window which appears.

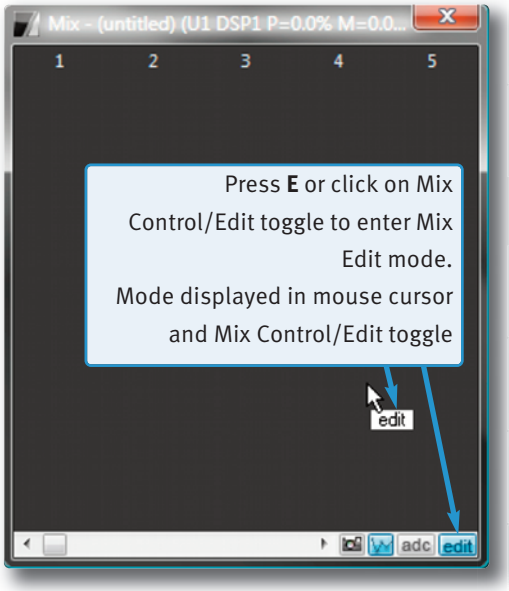


## Mixer Editing Tools

Seven of the tools in the Soundscape Toolbar can also be used for creating, and configuring the SSL Mixer's channels. While simple to use, they allow you to design almost any structure you can think of.




These Tools only work when Mix Edit mode is active. To enter Mix Edit mode, click on the mouse icon in the bottom left-hand corner of the Mixer window so that it displays **edit**, or simply press the ***E*** key. Details of how to use these tools can be found within the descriptions of their associated operations, located in this chapter. The keyboard shortcuts shown below can be used to assign each tool to the left mouse button:






**Create tool**: (***Alt+1***) Create new Strips in empty columns or new mixer elements in the User definable area of the Strip



**I/O Assign tool**: (***Ctrl+1***) Change I/O's for Inputs/Outputs and Sends, assign Strip to another DSP. The ***Ctrl*** key can be used to assign I/O on multiple channels – see *Assigning I/O on Multiple Mixer Strips* for details.



**Mute tool**: (***Ctrl+U***) DSP mute mixer elements or complete mixer columns



**Move tool**: (***Ctrl+M***) Move mixer elements inside Strip or mixer strips to another column (the Move vertical tool can also be used)



**Copy tool**: (***Ctrl+C***) Copy mixer elements inside Strip or mixer strips to another column (the Copy vertical tool can also be used)



**Delete tool**: (***Ctrl+D*** or ***Del***) Delete mixer elements or mixer columns



**Info tool**: (***Ctrl+I*** {letter i}) used to display information about mixer elements or mixer column

Four tools can be 'loaded' onto the mouse buttons simultaneously. Click on the tool of your choice with the mouse button you want to assign it to, and a black bar will be displayed below the tool's icon, to the left or right side according to the mouse button you used. You can select two further tools by holding down the ***Alt*** key while you click the tool icons; This status will be shown by a red bar under the tool. To use these tools hold down the ***Alt*** key while using the mouse button.

*NOTE: If the same tool is assigned to a mouse button and the **Alt** key + that same mouse button combination, then only the white bar will be shown.*

### Selecting multiple channels

Tools can also be used on multiple mixer columns simultaneously. To select multiple columns for editing, hold ***Shift*** then click on the main body of first and last columns in a range. You can add an subtract individual columns from your selection by holding ***Ctrl*** and then clicking on a column. Note that clicking on the Input or Output areas will not work.

By default, a selected range of columns will be deselected once a tool has been used. However, if the **Keep Column Selection** option in **Settings** / **Preferences** / **Mixer** is selected, selected columns will remain selected until a new selection is made or an empty column is clicked on.

*Note that the Create and Info tools do not work on multiple columns. If multiple columns are selected when using the Info tool on a column, the multi-selection will be cancelled and only the selected column's info page will display. Other tool operations also change slightly with multiple channel strips selected, as described within the relevant sections of this chapter.*

## Structuring the Mix Window

The key to understanding the Mixer is simple: you can have virtually any type of configuration you want, provided that you do not exceed available DSP processing power. The number of mixer strips, EQs, sends, returns, peak meters or faders and their positions in the mixer are completely user-definable. In fact, every part of the mixer is a plug-in element, just like a plug-in DSP effects algorithm.

The only fixed parameters are the number of streams, number of buses and number of physical inputs and outputs available per card. Let's define what we mean by 'inputs and outputs, 'buses' and 'streams':

**Inputs and Outputs:**

Physical connections to audio devices outside the host computer, such as microphone preamps, mixing desk, stereo mastering recorder and so on. They may be called 'physical' or 'external' inputs and outputs and should not be confused with the 'input elements' and 'output elements' that will be described later in this chapter. In particular, while input elements and output elements may be assigned to an external input or output, they may just as well be assigned to a bus or stream...

**Buses:**

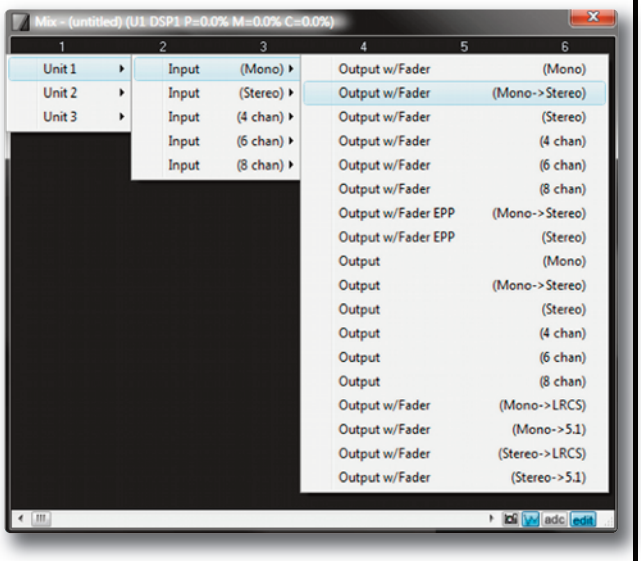
Audio paths/connections that can be used to route audio signals between different parts of the SSL Mixer, for instance between a send element and an input element. For the purpose of this manual, 'buses' only exist within the SSL Mixer.

**Streams:**

Audio paths/connections that can be used to route audio signals in either direction between the SSL Mixer and Windows applications, such as MIDI+audio sequencers or virtual instruments.

### Creating Mixer Strips

To create a mixer strip, activate the Mix Edit function (by clicking on the **edit** or mouse symbol in the bottom right-hand corner of the Mix window or pressing the ***E*** key), assign the Create tool to a mouse button, and then click with that mouse button in the area of the window in which you want the Mixer Strip to appear.




Now use the drop-down menu which appears to define the type of channel you want to insert:

- In the first menu, select the required **Unit** (if multiple Units are available);
- In the next menu, select the number of input channels;
- In the next menu, select the number of outputs and choose whether to include an output fader, or an EPP (Equal Power Panning) fader.

*For a more detailed description of the I/O options, please see 'Mixer Strip Input/Output configurations'.*




A mixer strip will appear with the default inputs and outputs (1-2 U1), a balance control slider with centre button, a pre-fader stereo peak meter, input and output mute buttons, a solo button and a fader.

There is also a blank space at the bottom of the mixer strip that can be used to name and colour-code the Strip.

## Deleting Mixer Strips

To delete a Mixer Strip, activate Mix Edit (by clicking on the **edit** or mouse symbol in the bottom right-hand corner of the Mix window or pressing the ***E*** key), assign the Delete tool to a mouse button. Then click on the strip to be deleted and click **Yes** in the warning pop-up which appears.

*Note that you can select multiple mixer strips for deletion.*

## Moving Mixer Strips

To move a Mixer Strip, activate Mix Edit (by clicking on the **edit** or mouse symbol in the bottom right-hand corner of the Mix window or pressing the ***E*** key) and assign the Move tool to a mouse button, mixer strips can now be moved by clicking-and-dragging them.

*Note that you can Move multiple mixer strips*

If you Move a mixer strip to a location containing another strip, the strip which was in that location will shift to the right to accommodate the strip being Moved (as will any strips further right, until there is an empty strip which can be filled).

*When moving multiple mixer strips with gaps in between them, all Moved strips are effectively inserted to the right of the first Moved strip, with gaps preserved. The strips which were in between are shifted to the right of the Moved strips. For example, if you are using Mixer Strips 1-5 and move Strips 2 and 4 to the right, the new strip order will be: 1, space, 2, space, 4, 3, space, 5.*

## Copying Mixer Strips

To copy a mixer strip, activate Mix Edit (by clicking on the **edit** or mouse symbol in the bottom right-hand corner of the Mix window or pressing the ***E*** key) and assign the Copy tool to a mouse button. Mixer strips can now be moved by clicking-and-dragging them.

*Note that you can Copy multiple mixer strips*

If you Copy a mixer strip to a location containing another strip, the strip which was in that location will shift to the right to accommodate the Copied strip (along with any strips further right, until there is an empty strip which can be filled).

*When copying multiple mixer strips with gaps in between them, all new columns are effectively inserted to the right of the first new strip, with gaps preserved. The strips which were in between are shifted to the right of the new strips. For example, if you are using Mixer Strips 1-5 and Copy Strips 2 and 4 to the right, the new strip order will be: 1, 2, 2', space, 4', 3, 4, 5.*

## Appending Mixer Files to the Current Mixer

There are two ways to append Mixer files to the current Mixer:

Firstly, use the **File** menu's **Append Mix...** option and select the Mixer file to be appended using the standard Windows file navigation dialogue which appears. The imported Mixer Strips will be added to the right of the last Strip in the current Mixer.

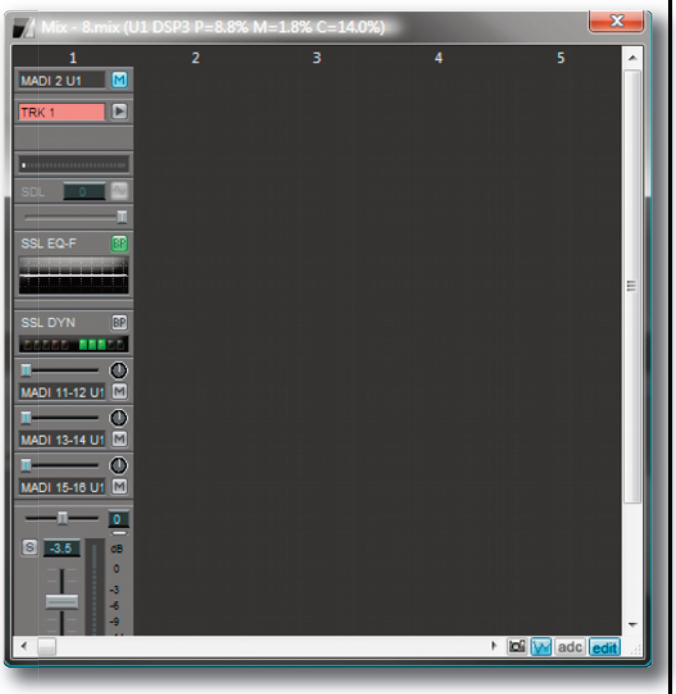
Alternatively, you can drag-and-drop straight from the File Manager: Simply locate and select the appropriate file in the File Manager and drag it to the location in the Mixer window where you want those Strips to be imported. A preview of the columns to be imported will be displayed while the mouse button is held down. When you release the button, the Mixer will shift the relevant Strips to the right in order to make space for the imported Strips.

## Placing Elements into Mixer Strips

Clicking with the Create tool in an existing mixer strip opens a menu displaying submenus for all mixer elements that are available to insert. The selected element is inserted as soon as the mouse button is released (or re-clicked).

There is no limit to the number of elements that can be inserted in a mixer strip (within the available DSP power). Even if there seems to be no space left in the mixer strip, existing mixer elements will automatically be moved to make space for a new one at the point where you click.

If you insert more elements than can be displayed in the SSL Mixer window, a vertical scroll bar will appear. It is also possible to resize the window.




## Signal routing in the mixer strip

The signal routing is always from the top to the bottom of a mixer strip. For instance, if you place a track insert below an EQ, this is the position at which audio will be recorded or played back (ie., if an audio sequencer records from this particular track insert, it will record the equalized signal).

## Using the internal buses

The Mixer has 64 internal digital busses for each hardware I/O unit installed on the host computer. These can be used to distribute or group signals that are to be sent to an internal effects processor or a master fader. Any output or send element can be routed to a bus and the input for any mixer strip can also be a bus instead of an external input. The same bus can be the input for as many mixer strips as required, allowing easy signal distribution in the Mixer.

## Basic Mixer Strip Structure

This diagram shows signal routing through a stereo input mixer strip.

The signal can be fed from an external input, sent internally from another mixer strip via a 'bus', or received from a Windows application via a streaming input.

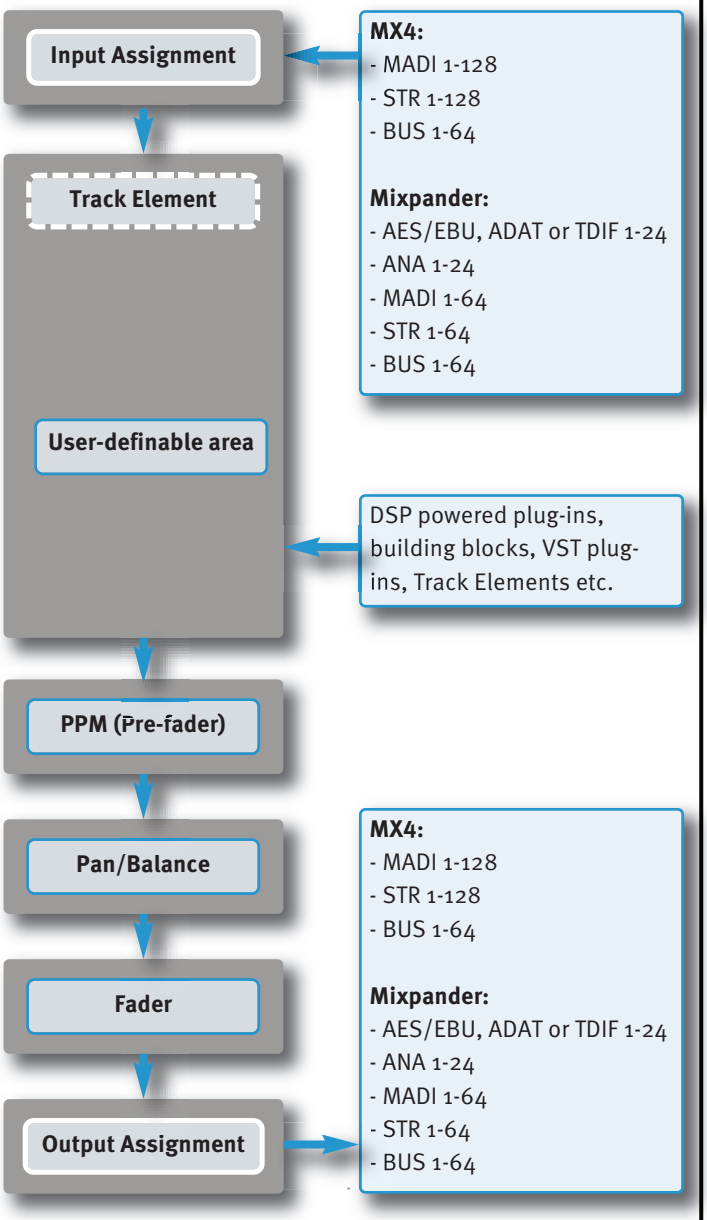
The Track Elements represent the 'Tape' Tracks, where the Audio can be routed to/from disk. The Track Elements can be freely positioned inside the user definable area, hence Pre or Post certain Plug-Ins. If a Track is located after a Plug-In, the processed Signal may be recorded to Disk.

Once the signal is in the strip, it can be mixed with a track output, equalized, delayed, its level adjusted and monitored, and then get sent to another mixer strip or output top be processed by 'plug-ins'.

The fixed peak meters are pre-fader, so they show signal even if the fader is down or if the mixer strip output is muted. Any level changes resulting from an inserted fader or EQ will be shown by the fixed peak meters.

The numbering and naming of the available inputs/outputs, buses and streams may vary depending on the hardware configuration; ie. The number and type of SSL audio cards installed in the host computer, the currently selected sample rate, the selected MADI format (56Ch and 64Ch) and High Speed (64-96kHz) modes for AES, ADAT and TDIF.

*Note that MadiXtreme routing is hard-wired – track 1 uses MADI input 1 and output 1, and so on.*

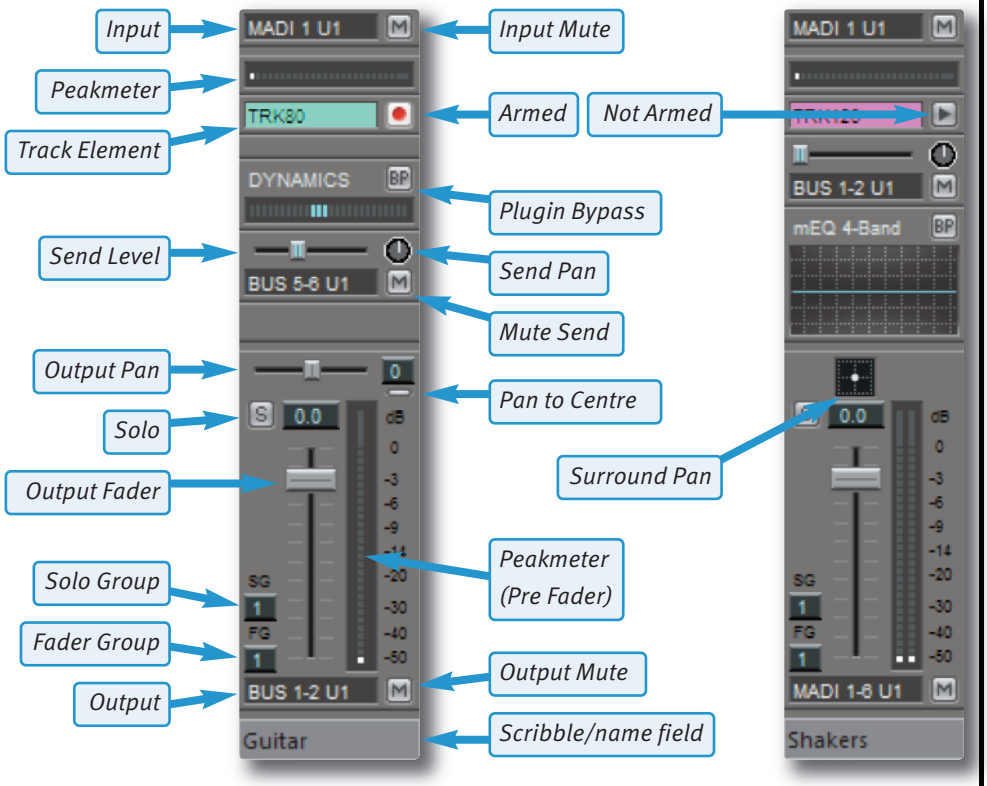


# Basic Controls

## Mixer strips

Mixer Strips in columns have the following standard controls and buttons:

- **Input** element on the top and an **output** element on the bottom, each with **mute** buttons
- **Scribble Strip** (name field) with definable Font and Background Colour
- On Outputs with Fader: a pre-fader **peak meter**, a **solo** button, Volume **Fader**, a **solo group assignment** box, and a **fader and mute group assignment** box.
- If the mixer strip has a stereo output, this output includes: a **pan** control (for mono in/mono to stereo out mixer strips) or **balance** control (for stereo in/stereo out mixer strips), and a **centre** button to easily reset the pan/balance to centre.

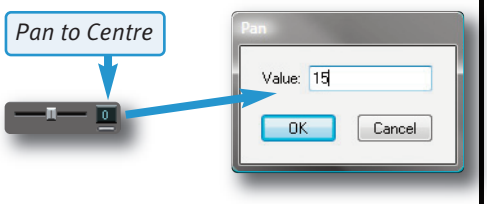



- If the mixer strip has an LRCS or 5.1 output: a **surround panner**
- On most plug-in elements: a **bypass** button.
- On sends, input and output elements: a **mute** button (and a pan/balance pot if the send is stereo).

## Output pan

Output pan and balance sliders can be clicked and dragged with the mouse. It is also possible to change the value in the box, to the right of the slider:

- Right or left-clicking in the box respectively increases or decreases the value in steps of 1.
- Right-clicking or left-clicking in the box and holding the mouse button down respectively increases or decreases the value continuously. The other mouse button can then be held down as well to speed up the process even more.
- Mouse Scroll Wheel: hovering over the value box or the slider when the Mixer Window is active allows value increase with scroll wheel Up and decrease with scroll wheel down.
- Double-clicking in the value box calls up a dialogue box where the required value can be entered.
- Clicking on the centre button below the value box resets the value to 0=centre



## Fader and mute groups

Each output element with a fader can be assigned to one of 99 fader groups, and to one of 99 solo groups. Group assignments are regardless of the output type, (mono, stereo, mono to stereo, 4, 6 or 8 channels, LRCS, or LRC-LsRs)

*NOTE: A single mixer strip can contain several output elements, each assigned to different fader and solo groups.*

The fader group and solo group assignments can be selected in several ways:

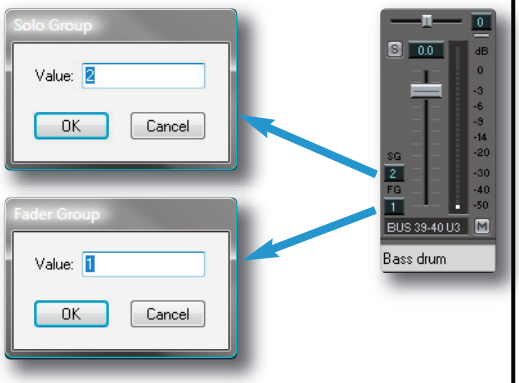
- Right-clicking or left-clicking in the relevant group assignment selection box respectively increases or decreases the value in steps of 1.
- Right-clicking or left-clicking in the selection box and holding the mouse button down respectively increases or decreases the value continuously. The other mouse button can then be held down too, to further speed up the process.
- Mouse Scroll Wheel: hovering over the group box and increase with scroll wheel Up, decrease with scroll wheel down
- Double-clicking in a selection box opens dialogue box where the required fader group number or solo group number can be entered with the keyboard.

### Faders

Whenever several output faders are assigned to the same fader and mute group, clicking and dragging any single one of them with the right mouse button will move all faders in the group .

The grouped faders keep their original position relative to each other while they are moving, until one or more of them reach minimum or maximum level and cannot move any further.

Even then, moving back will restore its relative position within the group, provided the mouse button has not been released.




### Mute buttons

The mute buttons for grouped faders are also linked. Right-clicking on a mute button will cause all the mute buttons in the same fader and mute group to switch to the same muted/unmuted status, regardless of their status before. The mute buttons can be used individually, regardless of their fader group assignment, by clicking them with the left mouse button.

## Solo groups

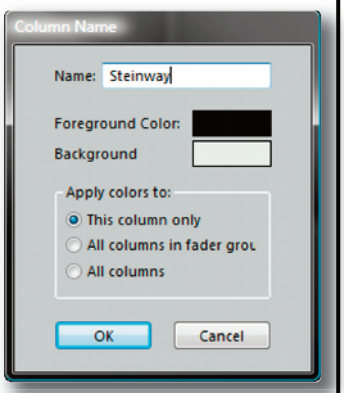
- Left-clicking any inactive (grey) solo button solos the corresponding output and silences all other outputs that have the same solo group assignment.
- Any previously soloed output inside the same solo group is dropped out of solo mode and silenced, unless the ***Ctrl*** key is used as described below.
- The solo button turns bright red (**Solo Listen**), all other solo buttons in the same group switch to a red **S** on grey background (**Solo Muted**).
- Left-clicking again on the active (red) solo button deactivates Solo Mode for this entire group.
- Right-clicking an inactive solo button solos the corresponding output and all the outputs that have the same fader and mute group assignment.
- It also mutes all the outputs that have both a different fader and mute group assignment and the same solo group assignment as any one of the soloed outputs.
- Any previously soloed output which has a different fader and mute group assignment and the same solo group assignment is dropped out of solo mode and silenced, unless the ***Ctrl*** key is used as described below.
- The solo buttons of soloed outputs turn red, and the solo buttons of silenced outputs display a red **S** on a grey background.
- Right-clicking on an already active solo button deactivates it, and also deactivates any other active solo button that has the same fader and mute group assignment.

### Solo buttons and the *Ctrl* key

If the ***Ctrl*** key is held down while left-clicking or right-clicking an inactive solo button, everything works as described previously except that any outputs that are already soloed remain soloed. The ***Ctrl*** key has no effect when left-clicking or right-clicking an already active solo button.

## Scribble Strips

Double-clicking on the Scribble Strips (Column Name Field) at the bottom of a mixer column opens the Column Name dialogue box .




This function cannot be accessed while in **Mix Edit** mode.

You can define:

- The **Name** of the Channel
- The Label Text Colour (**Foreground Colour**)
- The Label Colour (**Background** Colour)
- The choice of colours can be applied to the current column only, columns inside the same fader group or all Columns in the Mixer.

The Colours are defined by standard Windows Colour Dialogues which open when you click on the colour pads for foreground or background colour.




Using colour coding for text and labels really helps to maintain an overview and allows quick identification of grouped columns, especially when you start to build bigger Mixers and explore the endless possibilities of DSP based grouping.

## DSP Resources and DSP Auto Routing

It is possible to 'DSP mute' mixer strips or mixer elements using the Mute tool (ie. 'turn the Hardware off'), and unlike what happens in a hardware only digital mixer, in the SSL Mixer these muted mixer strips or elements do not consume DSP power anymore. This means that you can construct a Mixer containing duplicated strips with different parameter settings, for easy A/B comparison. Or you could build a Mixer that has a higher processing power requirement than the DSP(s) can provide, and just activate the elements that you need at any given time.

This way, if you need to use more effects than available DSP power allows, you can adjust an effect in real-time, record the output to a track, and then mute the real-time effect to save processing power for other purposes.

*NOTE: Pressing the BP button (bypass) for an element does not reduce the DSP power consumption, as the element is still active (just bypassed) and instantly available. Only DSP muting the element with the Mute tool will reduce the DSP power used.*

Each mixer element consumes a certain percentage of DSP resources and the overall percentage for processing **(P)** and memory **(M)** is shown in the Mixer window's title bar. For example, for an MX4 a single mono mixer column with output fader is around **(P)0.1% (M)0.3%**, the 4 band mono multi EQ is **(P)1.1% (M)0.3%**, a mono peak meter is **(P)0.1% (M)0.1%**.

More complex elements use more DSP power, such as for example the stereo Chorus/Flanger from the Audio Toolbox plug-in which uses **(P)2.1% (M)1.2%**. If you DSP mute an element, you will see a reduction in DSP power usage, as long as there is no 'DSP masking' taking place (another instance of the Plug-In is active and still needs to run some global plug-in code).

The DSP power percentage that is reported as being used for each element is the maximum percentage that this element will use. In reality, there are several parameters that are important to the DSP, for example processing time, and program and memory requirements. This means that when two elements are combined, the overall DSP power requirement may be less than the sum of both percentages (ie., one element may need more memory and the other may take more processing time, but with less memory used). In this case one algorithm is 'masking' the power requirement for the other.

*With multiple cards, the percentages are shown for each card individually.*

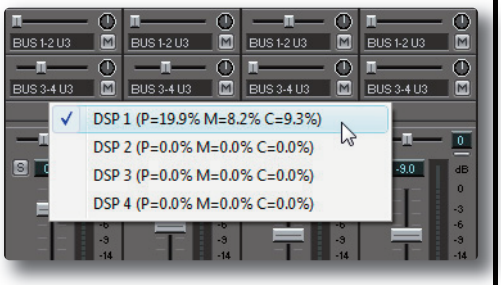
The **(C)** percentage shows the connection resources usage. Since each DSP has a fixed amount of Audio Paths connected to the main routing infrastructure of the hardware, connections are used by:

- Streams coming from or going to the PC (also for VST Plug-Ins)
- Physical Inputs or Outputs connected to the 'outside' world
- Internal Busses that need a path to another DSP

## Assigning a Mixer Strip to a specific DSP

*NOTE: This section assumes that **Mix Edit** mode is selected (press **E** on the keyboard to switch between Mix Edit mode and Mixer Control mode) and that the I/O Assign tool has been selected.*

Clicking in an empty area of a mixer strip with the I/O Assign tool brings up a list of the available DSPs for the Unit/card and current resources usage. This list is visible for as long as the mouse button is held down.




The ticked entry indicates the DSP Core that the mixer strip is currently assigned to (DSP 1 in this example). The DSP assignment can be changed by choosing another DSP before releasing the mouse button.

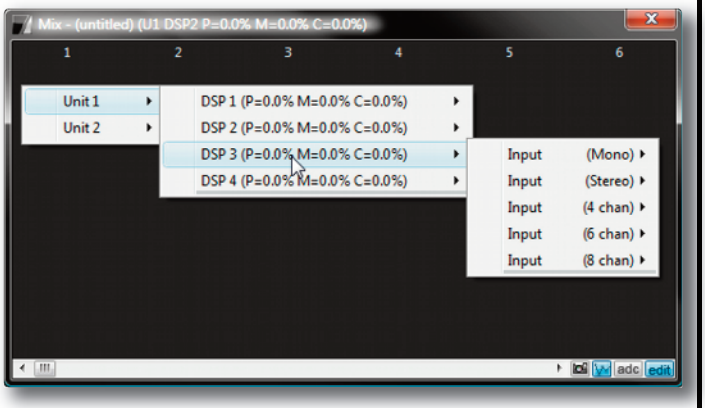
Note that multiple

### Mixer DSP Auto Routing

Auto routing can be enabled or disabled by ticking or unticking the **Auto route** item under **Settings** / **Preferences** / **Mixer**.

**Mixer auto routing enabled**

If **Auto route** in the **Settings** / **Preferences** / **Mixer** menu is ticked (auto routing enabled), DSP resources are allocated automatically.




**Mixer auto routing disabled**

If **Auto route** in the **Settings** / **Preferences** / **Mixer** menu is unticked (auto routing disabled), DSP resources need to be allocated manually when building a Mixer.

When **Create tool** is used to build a new mixer column, an additional drop down menu lists the available DSPs with their current resource usage values.

The new mixer column will use the DSP selected (highlighted).

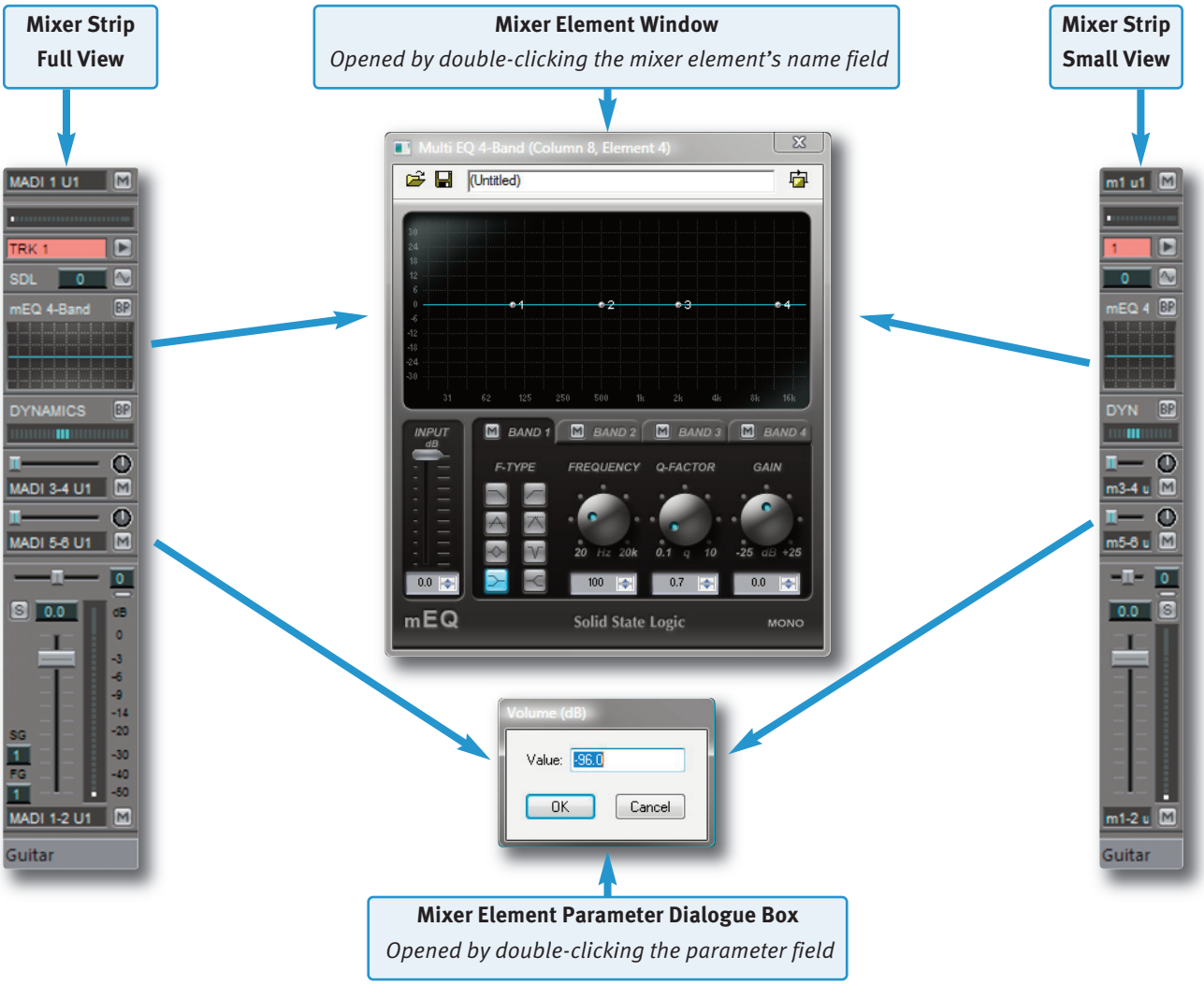
## Mixer Views

The SSL Mixer has two display modes: **Full View** (wide) or **Small View** (narrow). Both options are available under the **View** menu or by pressing the ***X*** or ***Q*** key on the computer keyboard. There is also an option to have the SSL Mixer window **Always on top** under the **Settings / Mixer** menu.

At 1024X768 display resolution, in **Full View** mode, up to 11 mixer columns can be displayed across the screen. In **Small View**, up to 17 can be viewed, but in less detail.

In particular, the solo group assignment boxes and fader and mute group assignment boxes are hidden in **Small View** mode. Similarly, less parameters are displayed for certain mixer elements in **Small View** mode. However, in most cases, double-clicking the name field of a mixer element will open a new, dedicated window where all parameters can be viewed and edited, and presets, if applicable, can be loaded or saved.

Double-clicking a parameter field will open a dialogue box where the value for that particular parameter can be edited with text input on the keyboard.




*NOTE: Once a Mixer has been loaded, it remains active even if the window is closed.*

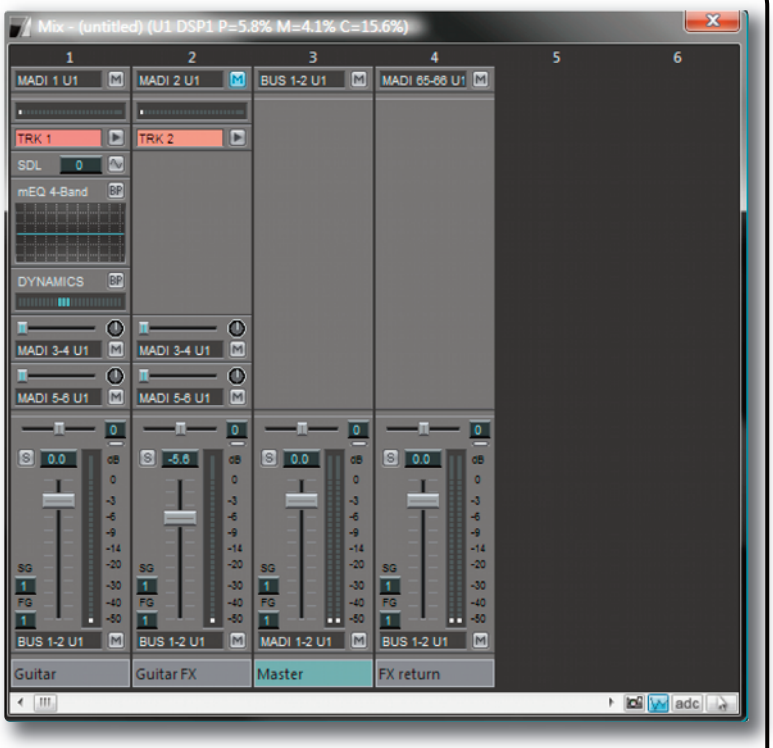
The Mixer contains 128 mixer columns, each contains one mixer strip that is 1 to 8 Channels wide. A horizontal scroll bar is provided for navigation across the Mixer window.

A vertical scroll bar appears whenever all the mixer strips cannot be viewed in the mixer window.

The Mixer on the right shows a simple structure with two stereo strips and an effects return strip, all bussed to a master mixer strip. Column 5 is empty.

Notice that only the first mixer strip has an EQ and Compressor, while both have a send element. This is an example of putting power where it is needed.

For example, if you want to send an EQ'd signal to an effects unit, drop an EQ in before the send. If you want a flat signal sent to the effects unit but want an EQ'd version sent to the master out, drop an EQ in after the send. If you need one auxiliary send in a strip and eight in another strip, just create them as required!



## Input/Output Identification

Depending on the SSL hardware that is used with the Mixer, the selection of available audio inputs and outputs will vary.

### MX4

Channel Inputs in the MX4 are named **MADI 1-64 (1-56)** for MADI Port A and **MADI 65-128 (65-120)** for MADI Port B, depending on the MADI I/O mode 64Ch or (56Ch).

The MADI Mode is selected under **Settings** / **I/O mode** / **MADI**, and has to match the settings of the connected MADI Devices (Converters, Consoles, Routers).

Internal Buses are named Bus 1-64 , the Streaming Channels (PC Software I/O's) are named STR 1-128.

*NOTE: When working at 96Khz, the number of MADI and Streaming I/O is reduced to 64.*

### MadiXtreme

MadiXtreme routing is hard-wired – track 1 uses MADI input 1 and output 1, and so on. There are therefore no routing changes which can be made by the user.

### Mixpander

Mixpander can handle 64 in/out from its expansion port, that can be connected to an SSL Alpha-Link or Soundscape iBox interface.

If connected to a legacy Soundscape 32 Unit, 26 inputs and 28 outputs can be used.

Mixpander also has 64 Buses (Bus 1-64) and 64 streaming channels (STR1-64).

On program start up, Soundscape detects which Unit is connected to the Mixpander and the I/O Labels will be named accordingly.

### SSL XLogic Alpha-Link (or Soundscape iBox 24/48/64 Models)

The SSL XLogic Alpha-Link and SSL Soundscape iBox 24/48/64 models offer a wide range of audio formats. Their inputs and outputs are identified by a string of letters for the output type (ANA for Analogue I/O, AES for AES/EBU I/O, MADI for... MADI I/O, ADAT for ADAT I/O, TDIF for.. TDIF, STR for Streaming I/O, BUS for Bus), one or two numbers (one for a mono input or output, two for a pair or larger group, eg. 9-16 for an eight-channel input element), and a unit number if there is more than one card present in the system.

*NOTE: When working at 96Khz, the number of connections to the Alpha-Link Unit and Streaming I/O is reduced to 32.*

### Soundscape 32 workstation connected to a Mixpander

The Soundscape 32 audio workstation offers 3 TDIF ports that provide a total of 24 i/o and are identified in the SSL Mixer as TDIF1 to 8, TDIF9 to TDIF16 and TDIF17 to TDIF24 inputs and outputs.

The AES/EBU stereo input connector is named DIGITAL AES/EBU IN1/2.

The two AES/EBU stereo output connectors named DIGITAL AES/EBU OUT1/2 and OUT3/4.

The two balanced XLR inputs are identified as the ANA1 and ANA2.

The four balanced XLR outputs are named ANA1, ANA2, ANA3 and ANA4.

*NOTE: When working at 96Khz, the number of I/O's is limited to 16 and Streaming I/O is reduced to 32.*

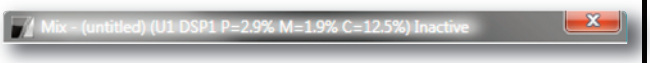
## Mixer Inactive Build

For each change that is made in Edit Mode, the DSP code needs to be updated – all of the DSP code may have to be restructured and spread differently across all DSP's. While this is happening, no further editing is possible. Depending on the amount of DSP Code that needs reloading (in other words, depending on the size of your mixer), this could take anything up to a second.

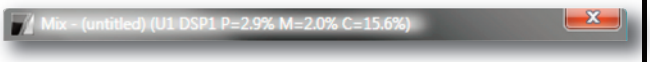
In order to speed up the building process, you can also use 'Inactive Build'. Here, the Mixer gets deactivated (no DSP Code loaded) and you can edit your structure, add Strips and elements instantly. Once you are done, just activate this Mixer and all of the DSP Code is loaded in one go.

If native mixer elements such as VST plug-ins are used, native processing is also stopped while the Mixer is being edited.

To use Mixer Inactive Build, hold the ***Shift*** key down while performing the required edits. The title bar will display **Inactive** after the **P**, **M** and **C** values, and the changes you make will be reflected instantly on screen.



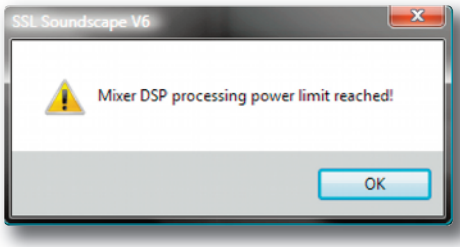

The ***Shift*** key can be released before the last edit is carried out in order to reactivate the edited Mixer.



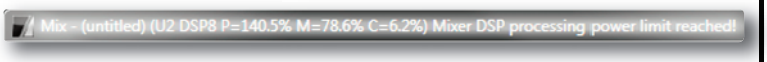

Alternatively, you can double click on the Caption bar to reactivate the Mixer.

### DSP power limit

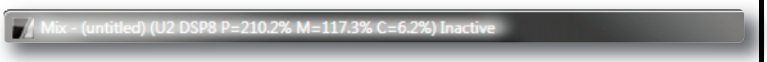
The **P** value reported in the Mixer window's title bar is the percentage of the available DSP clock cycles required to run the Mixer at the selected Sample Rate (taking the Varispeed setting into account). The **M** value is the percentage of memory resources required to run the Mixer. In most cases, if either of these values exceeds 100% when a Mixer is created, the following warning message is displayed.




After you click **OK**, the title bar will display the 'P' and 'M' values so that you can see which one exceeds the available resources, and by how much.




If only the **P** value exceeds 100%, the title bar will display a DSP Power Warning.




If the **M** value or both values exceed 100%, the title bar will inform you that the Mixer is inactive.




Muting or deleting some elements will reduce the values and reactivate the Mixer.

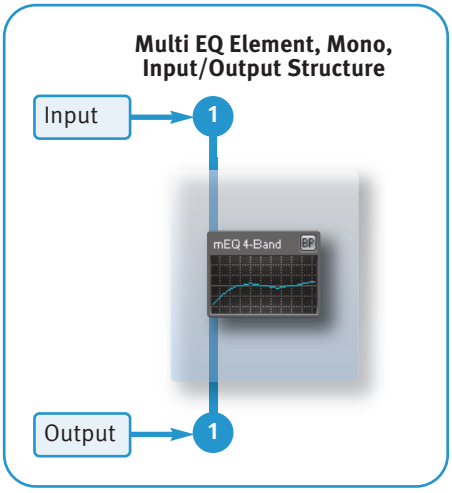
However, the **P=xx%** value is not always completely accurate, because some plug-in elements may report inaccurate processing cycles. This means that depending on the elements used, some Mixers can't reach or may exceed the **P=100%** value without any warning box appearing.

Also the Mixer DSP measures the actual processing cycles used for every sample, and warns if a real mixer processing cycle overrun occurs. If this is the case, a warning is given in the caption bar of the mixer window, and the mixer is muted for about half a second. The warning can also be reset by double-clicking the caption bar of the mixer window.

# Basic I/O and Routing Elements

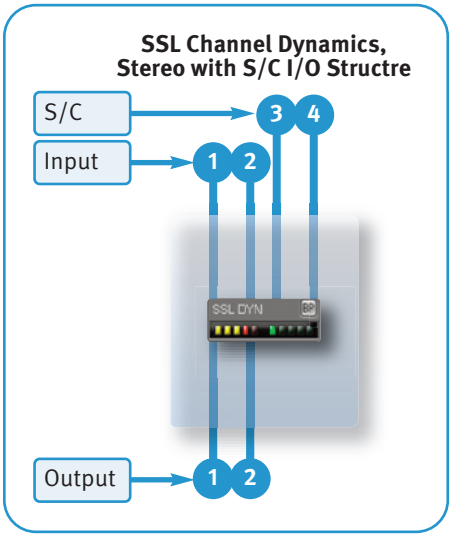
Each mixer element has at least one input and one output, and some elements have many more. These individual inputs and outputs come in three categories: 'fixed', 'assignable', and 'track element I/O', which is a mixture of both.

## Fixed inputs/outputs




The fixed inputs and outputs connect the various mixer elements within a mixer strip. In these diagrams they are indicated by numbers in the blue circles incremented from left to right. The example to the left shows a mono multi EQ mixer element with its single input and single output represented.

...and a stereo SSL Dynamics element with side-chain, as it appears within a mixer strip, and with its four inputs and two outputs represented.

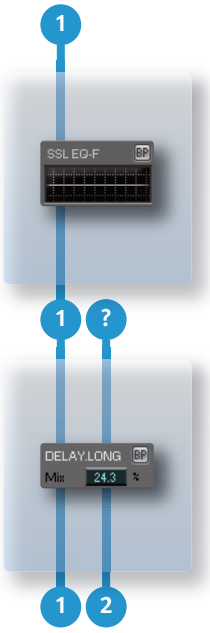



Fixed inputs and outputs always connect to an output or input with a matching 'digit'. For instance, an element with a single mono output, (output 1), will always connect to the left side input, (input 1), of a stereo element, as shown below. This type of configuration should be avoided, because the presence of an unconnected input can create problems:

**Incorrect configuration:**

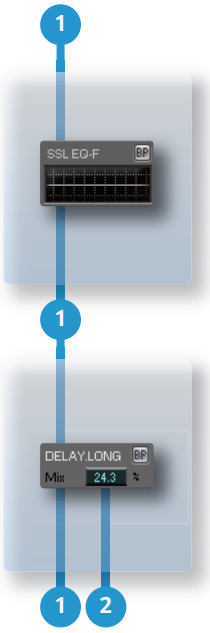
A mono mixer element (SSL Console EQ Mono), is connected to a stereo mixer element (Audio Toolbox Delay).

Input 2 of the stereo delay is unconnected. This could create audible problems and erratic routing.




**Correct configuration:**

A mono mixer element (SSL Console EQ Mono), is connected to a mono to stereo mixer element (Audio Toolbox Delay Mono->Stereo)

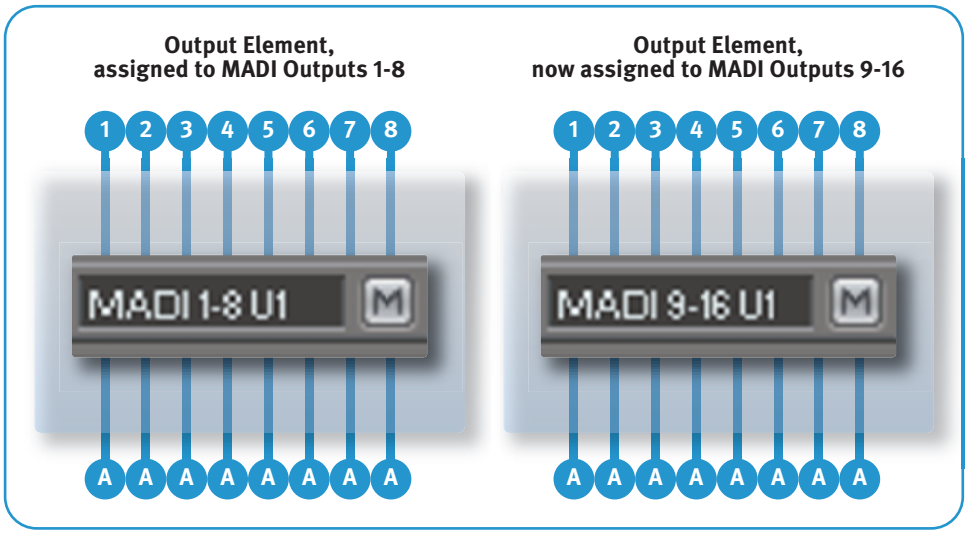


It is important to take this into account when building a Mixer, since the two configurations shown above would look the same within a mixer strip. If you need to check the input/output configuration of an existing mixer element, click on it with the Info tool in **Mix Edit** mode. A window containing the relevant information will be displayed.

*NOTE: The principles outlined in the examples above using mono, stereo and mono to stereo mixer elements also apply when dealing with mixer elements that have a higher number of inputs and outputs. For example, combining a 6 in/6 out input element and an 8 in/8 out output element within a mixer strip would result in an incorrect configuration, because inputs 7 and 8 of the output element would be left unconnected.*

## Assignable inputs/outputs

The assignable inputs and outputs connect mixer elements across mixer strips, to the external inputs and outputs of the MX4 or Mixpander, or to the host PC via the 'streams'. In our diagrams they are not numbered. Instead they are indicated by an 'A', meaning that their routing can be determined freely using the I/O Assign tool. The individual input(s) of input elements and individual output(s) of output or send element(s) are assignable in this way.



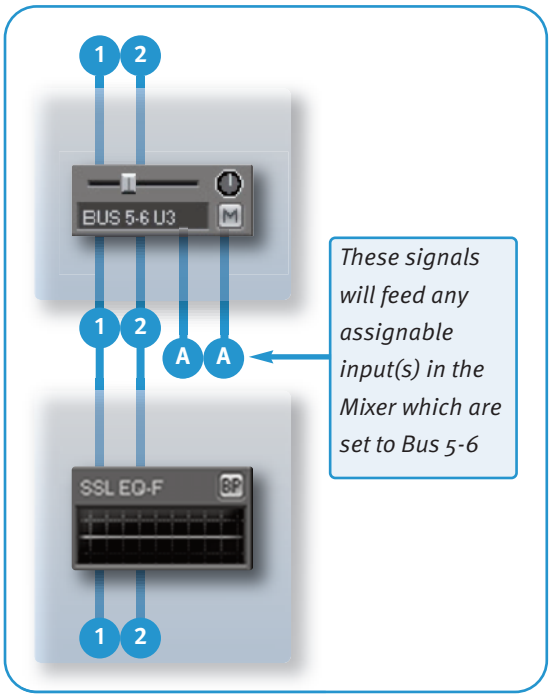

The example above shows an 8 in/8 out output element. Initially, the element's outputs are assigned to outputs 1-8. The I/O Assign tool is then used to change the element's output assignment to outputs 9-16.

As well as transmitting the audio data they receive via their assignable output(s), incl. any volume and pan/balance settings and the mute status, send elements and output elements also let the original data pass through to any element placed below them, (ie., unaffected by these settings and regardless of mute status).

These signals will feed any assignable input(s) in the Mixer that is (are) set to BUS 5-6.

In the example to the right, the data received at inputs 1 and 2 will be output :

- Via buses 5 and 6 (assignable outs shown as 'A')
- Original signal will reach the multi EQ element placed below the send



## Assigning I/O on Multiple Mixer Strips

Any Mixer Element that contains an I/O assignment area (Inputs/Outputs/Sends), can be changed on multiple selected columns simultaneously. Using the I/O Assign tool on any element within the current column selection will change the identical* elements on all selected columns to the chosen I/O value.

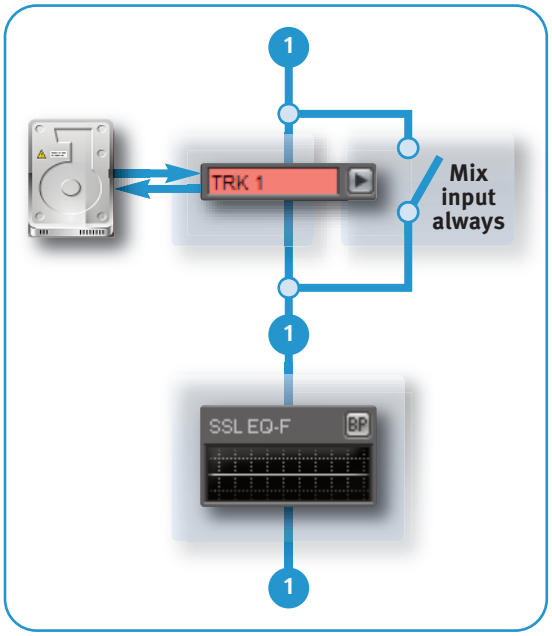
Pressing and holding the ***Ctrl*** key while clicking on a certain element inside the column selection will change this element to the chosen I/O, and then decrement this value on selected columns to the left and increment it on selected columns to the right.

**IDENTICAL ELEMENTS:*

- *The Mixer only considers elements to be identical across columns if they are using the same element type and their visual vertical position inside the Mixer strip is vertically overlapping or the same.*
- *If two of the same element type are visually overlapping (ie. 2 Send Elements), the top element will be changed.*
- *Mixer strips, I/O's and Sends with a different channel count or routing layout (ie.. Mono to Mono, Mono to Stereo, Stereo Send, Stereo EPP Send) are not considered to be identical, regardless of their vertical visual position.*

## Track Elements

The Track Elements are a mixture of fixed and assignable I/O's. Track Elements always connect to Audio Tracks in the Arrange window (fixed), however the Track they are connected to is assignable.



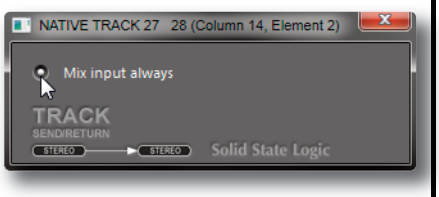

In Record mode (Track is armed) the Input of the Track is always routed to the disk and from there back to the Track Output (independent of the selected **Mix Input Always** status described below).

When an **Auto Punch In** or **Punch In/Out** is set in the Arrange Window during recording, the Track Element at Punch-In will switch from Disk Playback present at the Track Output to the Track Output being directly fed from the Track Input (if Mix Input Always is not checked).

This makes it possible to use a Bus/Tape Style Artist Monitoring and preview an overdub while recording it.

## Mix Input Always

In Play mode (Track not armed) the Input of the Track (1) is available to be mixed with the audio coming from disk. To do this, double-click on Track Element and check **Mix Input Always**) at the Track's Output.




The surround of the send's track arming icon will go blue to indicate that **Mix Input Always** is active.

*NOTE: Track Elements as well as 'native' VST Plug-Ins are using streams from the SSL Card to exchange audio between the DSP's and the PC's Host CPU. While you can select a certain Stream number for I/O and Send Elements, Track Elements and VST Plug-Ins will dynamically be assigned to unused streams (as long as there are any left) automatically. If no streams are left a '**stream exhausted**' error message will appear.*

## Mixer Strip Input/Output Configurations

When you select the Create tool and click in an empty mixer slot in Mix Edit mode, a menu appears, prompting you to create a new mixer column. If you have several units in your system, the menu will let you choose the unit (ie., the DSP and actual inputs/outputs) that the mixer column will be created for.

Several mixer column configurations are available which all include 'built-in' input and output elements.

The output element always has a mute button, and may have a fader, pan or balance control, peak meter, and solo button.

An EPP (Equal Power Panning) output can be selected for stereo and mono to stereo outputs.

### Mixer column input/output configurations using mono and stereo:

- For **mono in - mono out**, the signal just flows through the mixer column, only affected by the user defined settings of the various mixer elements, which also should be mono.
- For **stereo in - stereo out**, the respective level of the left and right signals is controlled by the balance setting (available if there is a fader). Stereo mixer elements would be inserted in the signal path as required.
- For **mono in - mono to stereo out**, the signal path through the mixer column is mono, and the signal is split into stereo before the pan control.

For **mono in - stereo out**, the input is mono, but the input signal needs to reach both inputs of a stereo output element. This mixer column configuration should only be created in order to use a stereo process or effect on a mono source (eg., a mono to stereo chorus or reverb).

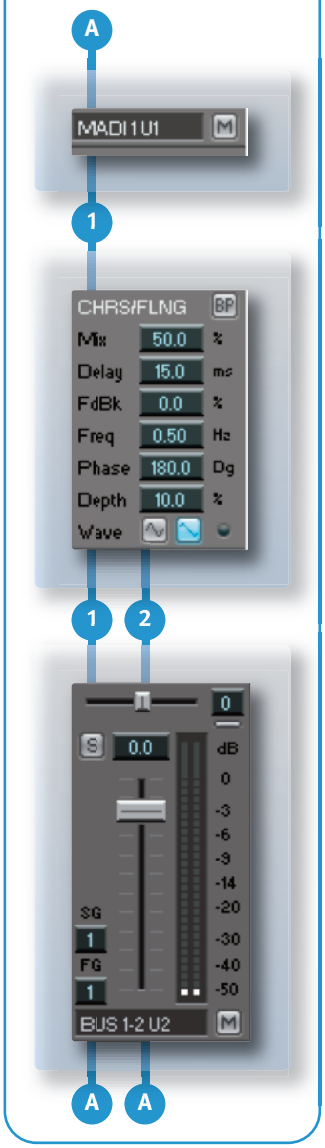
In fact, if you use this configuration without inserting a mono to stereo process in the mixer column, then the input element will not be connected to Input 2 of the output element, and this should be avoided as it can have undesirable effects (the right hand side channel of the output element may pick a signal from elsewhere in the Mixer).

In the example on the right, a mono signal reaches the mixer column via the MADI 1 input.

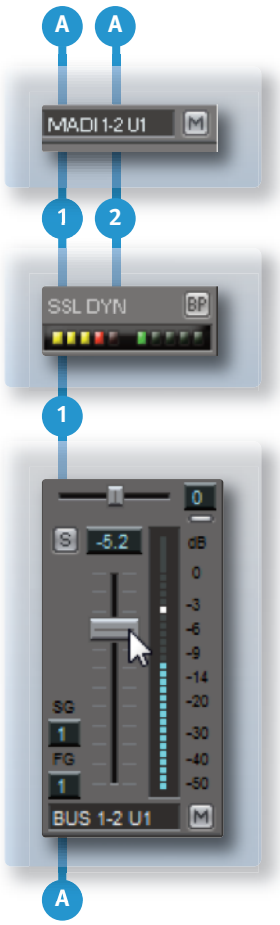
A mono to stereo Chorus effect is applied.

The chorus's stereo pair of outputs connects as required to the stereo output element.

**mono in – stereo out mixer strip connection diagram**



**Stereo in – mono out mixer strip connection diagram**




For **stereo in - mono out**, only the left side of the stereo input will be connected to the mixer column's output.

This configuration can be used to insert a 2 in/1 out mixer element such as a mono compressor with side-chain input.

The main signal will reach the compressor via MADI input 1, the key signal will reach the compressor via MADI input 2.

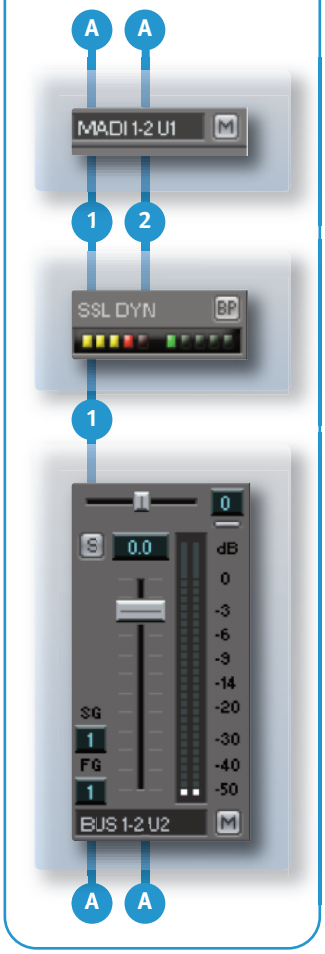
Only the main signal will reach the compressor's single output.

For **stereo in - mono to stereo out**, only the left side of the stereo input is connected through the mixer column, but it splits into stereo at the pan control.

This configuration, just like the stereo in - mono out configuration described above, can be used to insert a 2 in/1 out mixer element such as a mono compressor with side-chain input.

The only difference is that the output can be panned left or right as required with the mono to stereo output element.

**Stereo in – mono-to-stereo out mixer strip connection diagram**



## Stereo Balance and Panning

Stereo and mono to stereo outputs can be selected with Equal Power Panning (EPP menu options).

In this case, when the balance or pan control is moved to one side, the level for that side is gradually increased by up to 3dB to compensate for the overall loss of power from the other side.

## 4 in - stereo out

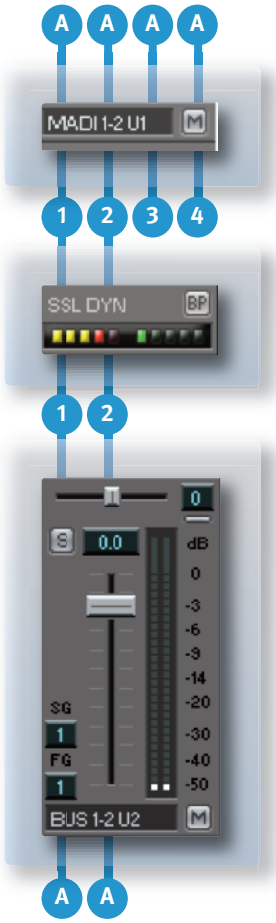
This configuration would be used to insert a 4 in/2 out mixer element such as a stereo compressor with side-chain input.

The main signal must reach the compressor via MADI inputs 1-2,

the key signal (or Sidechain) must reach the compressor via MADI inputs 3-4.

Only the main signal will reach the compressor's outputs.

**4 in – stereo out mixer strip connection diagram**



## Surround Mixing I/O Configurations

For **mono in - mono to LRCS out**, the signal path through the mixer column is mono, allowing mono processes or effects to be applied to the signal.

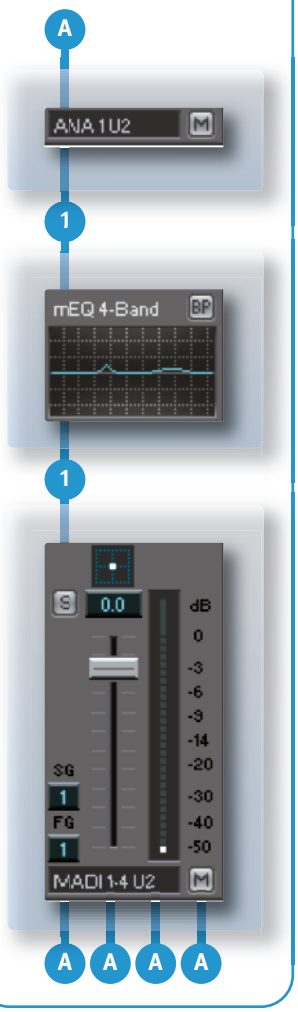
The mono multi EQ in the example, is applied before the eq'd signal is positioned in the sound field using the surround panner.

A **stereo in - stereo to LRCS out** configuration is similar to the mono in- mono to LRCS out configuration, except that the signal path through the mixer column is stereo, allowing stereo processes or effects to be applied to the signal.

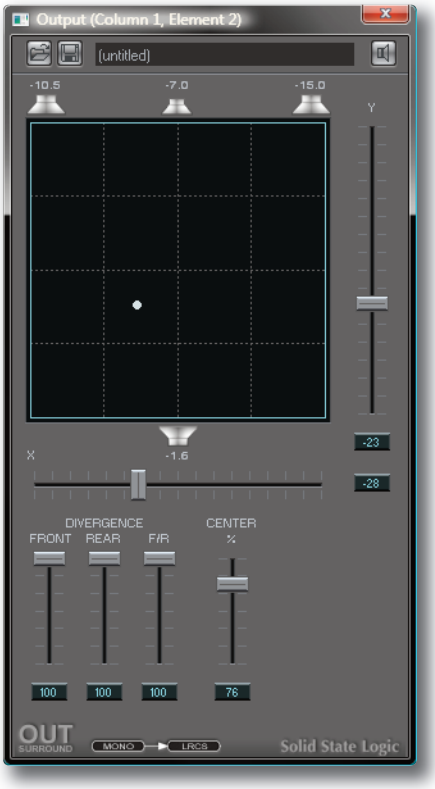
Routing in this example: In this example the output routing of the LRCS signal is:

- MADI 1: L
- MADI 2: R
- MADI 3: C
- MADI 4: S

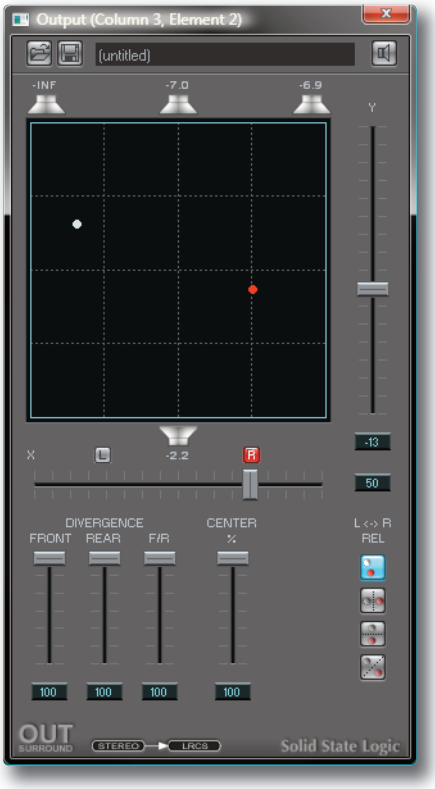
**mono in – mono-to-LCRS mixer strip connection diagram**




**mono to LRCS panner**




**stereo to LRCS panner**



The **mono in - mono to 5.1 out** configuration is similar to the mono in - mono to LRCS out configuration, except that the output element has six individual outputs for 5.1:

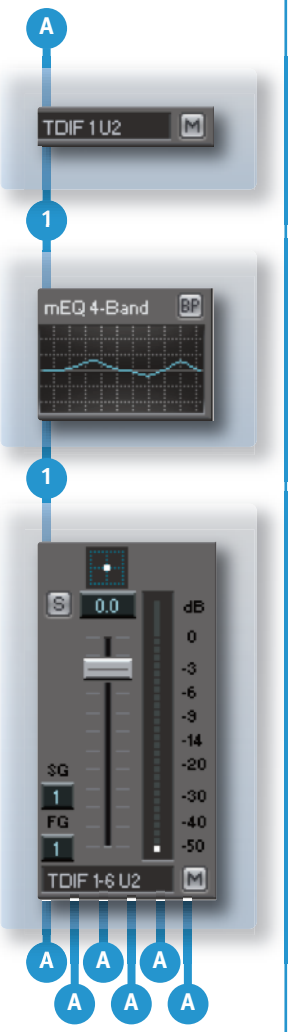
- L
- R
- C
- LFE
- Ls
- Rs

The **stereo in - stereo to 5.1 out** configuration is similar to the mono in- mono to 5.1 out configuration, except that the signal path through the mixer column is stereo, allowing stereo processes or effects to be applied to the signal.

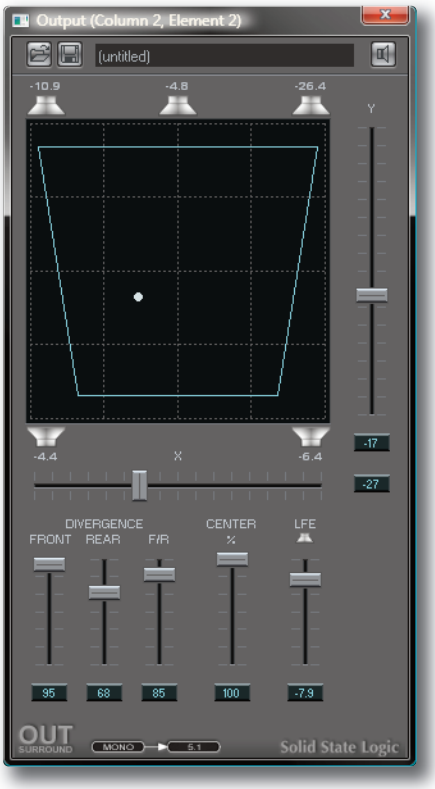
Routing in this example:

- MADI 1: L
- MADI 2: R
- MADI 3: C
- MADI 4: LFE
- MADI 5: L S
- MADI 6: R S

**mono in – mono-to-5.1 mixer strip connection diagram**




**mono to 5.1 panner**




**stereo to 5.1 panner**



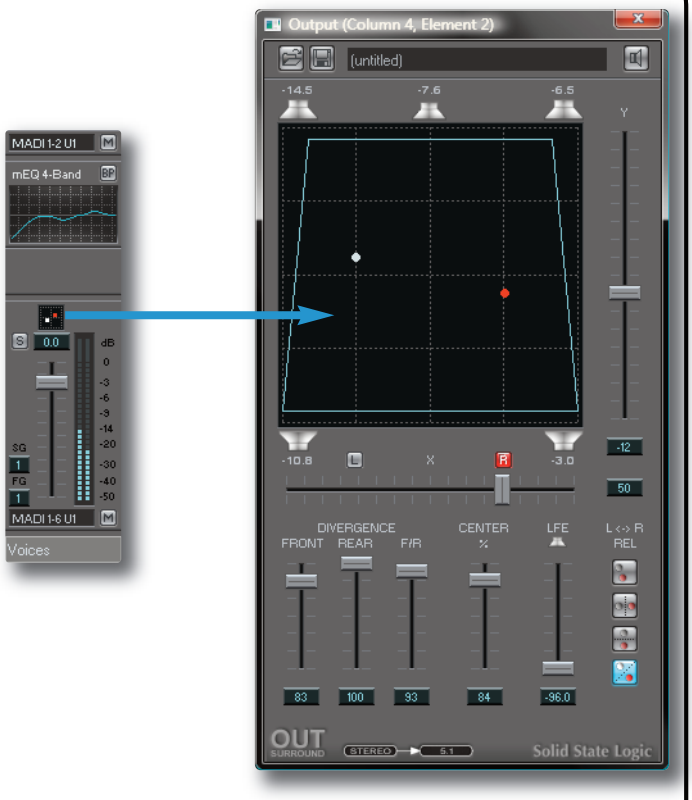
For mixer columns with a mono or stereo to LRCS or 5.1 output, double-clicking the output element just next to the small surround panner, or in the output assignment box will open the Surround Panner window.

The panner features individual pan dots for the Left (white or mono) and Right (red, stereo panner only) input signal(s).

These dots can be moved across the sound field using the left and right mouse buttons respectively, by clicking and dragging or by clicking at the required position in the sound field.

The pan dots can be moved either in the small panner in the mixer column or in the large panner window.

In the large window, it is also possible to use the **X** and **Y** sliders, or the **X** and **Y** value boxes. For surround panners with a stereo input, L and R buttons, positioned above the X fader, indicate whether the Left or Right signal panning is currently controlled by the X and Y faders.




The **Front**, **Rear** and **F/R** (Front/Rear) **divergence** controls allow the sound to blend gradually, respectively between the front speakers (as if the front speakers were brought closer to each other), rear speakers (as if the rear speakers were brought closer to each other) or front and rear speakers (as if all the speakers were brought closer to each other). Higher values create more separation.

The **Centre** Speaker control allows centre panned sound to be gradually spread across the left and right front speakers.

The **LFE** (dB) level control (for 5.1 output panners only) determines the level of the signal routed to the LFE channel (pre output fader). Clicking the 'Speaker' icon above the **LFE** fader will mute/unmute the LFE output channel (the LFE fader appears dimmed when the LFE channel is muted).

*NOTE: The panning mode is always 'equal power', regardless of panning position, divergence or centre speaker control settings. Muted speakers are not taken into account though!*

### X, Y, Divergence, Centre and LFE controls:

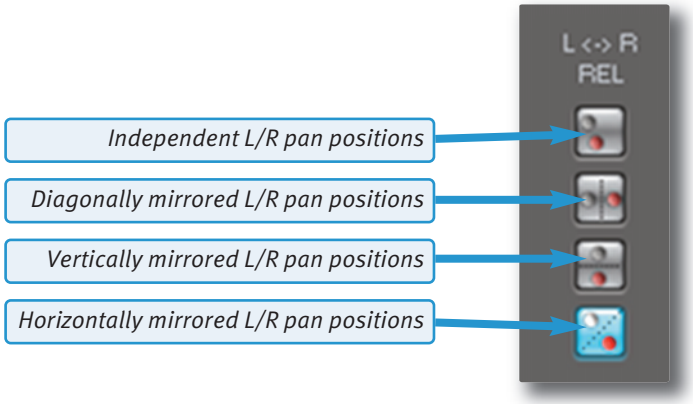
- The sliders can be moved either by clicking and dragging them or by clicking at the required position with the mouse
- Right-clicking or left-clicking in a value box respectively increases or decreases the corresponding value in steps of 1. Right-clicking or left-clicking in the selection box and holding the mouse button down increases or decreases the value continuously. The other mouse button can then be held down as well to speed up the process even more

### Speaker Icons:

- Clicking a speaker icon mutes/unmutes the corresponding speaker output
- Right-clicking a speaker icon solos/unsolos the corresponding speaker output
- The output level for each speaker is displayed in dB

### For stereo to LRCS and stereo to 5.1 output panners only:

- Left (L) and Right (R) buttons (positioned above the X pan fader) can be used to link the X and Y pan faders to either pan dot
- Four 'L<–>R Pan Relation mode' buttons allow the pan dot positions to be controlled independently, or allow the dots to be linked so that their positions are automatically mirrored horizontally, vertically or diagonally when moving either pan dot.




The output routing of the **mono to LRCS** and **stereo to LRCS** output panners is Left, Right, Centre, Surround, so that the left and right signals are on adjacent buses or outputs, in case just the stereo signal is required.

The outputs would be routed to four full bandwidth speakers, three across the front (L, C and R), and a rear surround speaker (S).

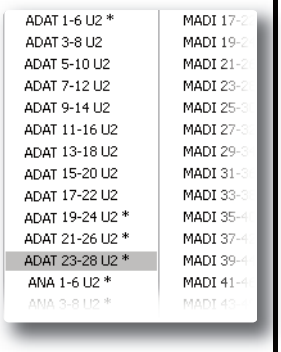
Similarly, the output routing of the mono to 5.1 and stereo to 5.1 output panners is Left, Right, Centre, LFE, Left surround, Right surround. For instance, if the output element is assigned to MADI1-6 U1, the output routing would be:

- L → MADI1 U1
- R → MADI2 U1
- C → MADI3 U1
- LFE → MADI 4 U1
- Ls → MADI5 U1
- Rs → MADI6 U1

## A note about 5.1 and ADAT/TDIF Outputs (Mixpander Only!):

When using a Mixpander card the component signals of a surround mix may be output via 8 channel ADAT of TDIF connectors. This way, the whole mix could for example be sent to an 8 track DTRS machine for mastering (a 5.1 mix could be output via the first six outputs, along with a stereo mix via the last two, so that all the material fits onto the same tape).

However, when the Track Assign tool is used to select an output for a mixer column with an LRCS or 5.1 output, the pop-up menu offers options that combine ADAT/TDIF and channels on different connectors (eg., the ADAT5-10 U2 option uses the ADAT A and ADAT B connectors).




Some of the options combine ADAT outputs with analogue outputs, or AES/EBU outputs with analogue outputs.

For these options, in the pop up menu, ADAT25, ADAT26, ADAT27, and ADAT28 in fact refer to ANA1, ANA2, ANA3 and ANA4.

For example, if you select ADAT23-28 U2 as the outputs of a 5.1 mixer column.

In fact this means that ADAT23 and ADAT24 will be used for the Left and Right signals, ANA1 will be used for the Centre signal, ANA2 will be used for the LFE signal, ANA3 will be used for the Ls signal, and ANA4 will be used for the Rs signal.

## Other mixer column input/output configurations

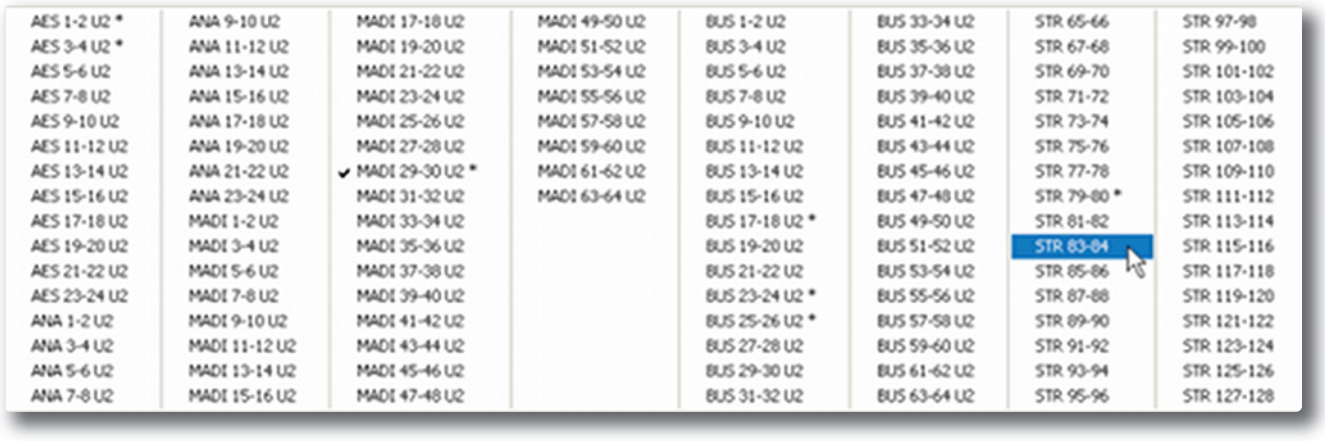
In addition to all that has been mentioned above, mixer columns can also be created with 4, 6 or 8 inputs and outputs. These could be used for example to send and receive multitrack submixes to and from 8 Channel (ADAT/TDIF) ports.

## Input and output selection

Each mixer column has at least one (built-in) user selectable input and output and may have any number of send elements. These can be set to use any of the physical inputs and outputs, internal mixer buses or PC audio streams.

In **Mix Edit** mode, selecting the Track Assign tool and clicking the input, output or send element will open the menu showing all possible routing options.

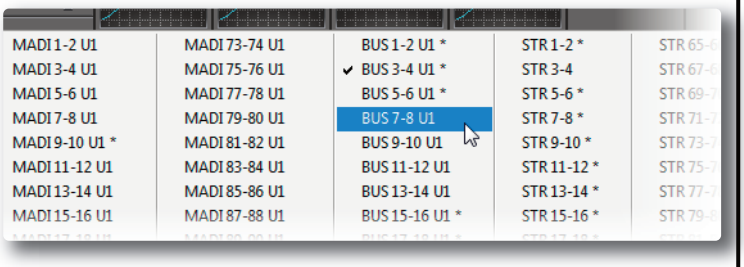
The example below shows the selection menu for a stereo input element with an SSL Alpha-Link MADI-SX connected to the Mixpander that the mixer column is running on.




*NOTE: The inputs and outputs, buses and streams that are already in use are indicated by an asterisk: (*).*

There are no restrictions to how you can use the inputs, outputs, buses or streams, and every signal routed to an output or bus is mixed in equal proportion (after the output or send fader). For instance, you could have 8 channels with MADI 1-2 U1 (or Bus 1-2) as the input, and Bus 3-4 as the output routing. You could then use different EQ settings to split the signal, process it with different effects (eg., for a multiband compressor) and recombine the signals at Bus 3-4. You could even select the same bus as the input and output of a mixer column, but be careful as you can have digital feedback this way!

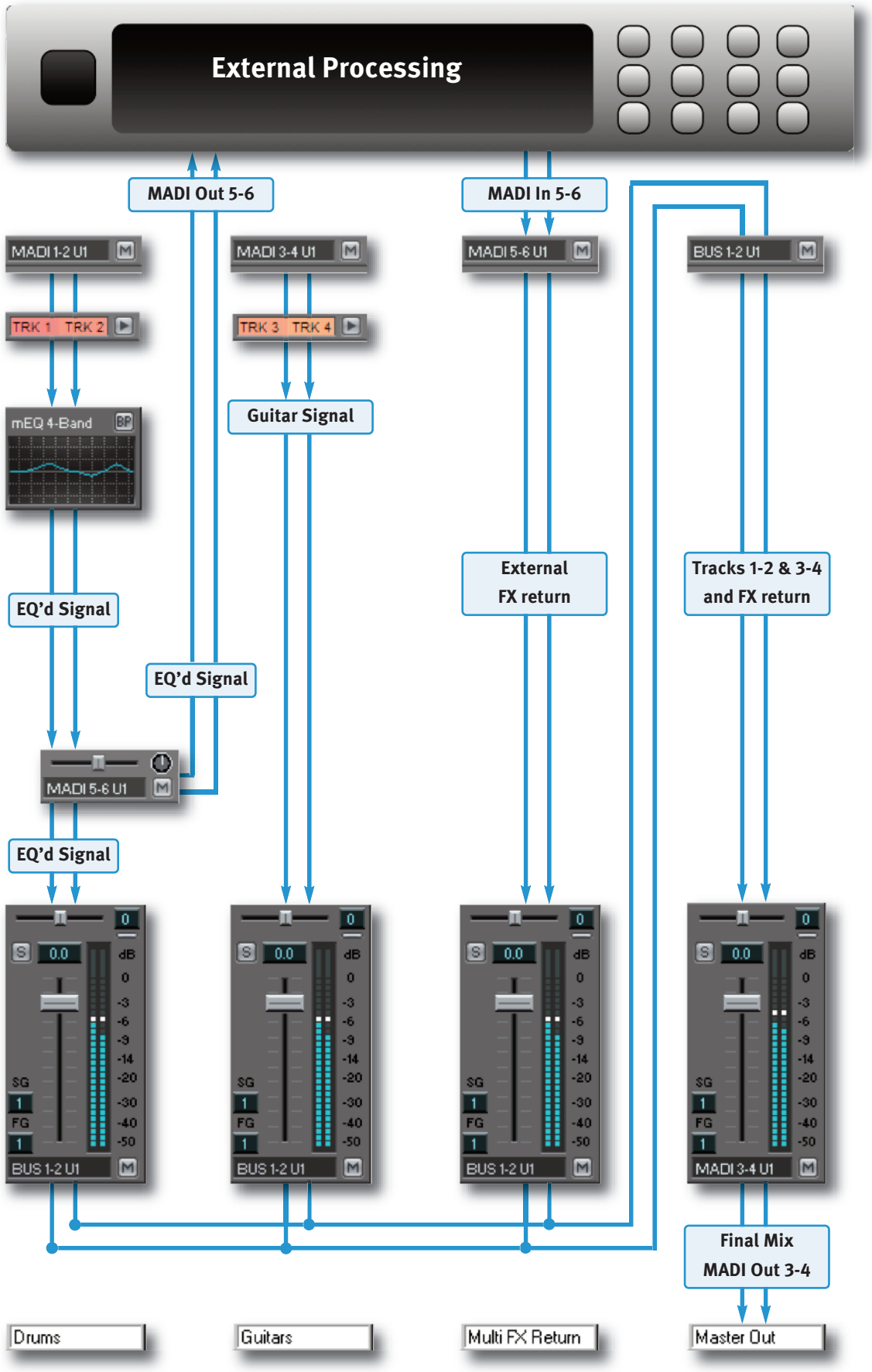
This example shows the menu for a stereo send, in a mixer column running on a MX4:




The streaming channels are automatically available to all audio programs. For example, they can be used as inputs and outputs in a MIDI + Audio sequencer..

*NOTE: If the output(s) from several mixer columns are sent directly to an external output (or group of outputs), it is not possible to view the combined output signal in order to check for overload. Therefore, it is advisable instead to route the mixer column outputs to a bus, and then to create an additional mixer column which has that bus as its input. This mixer column should have a fader with peak meter, for monitoring the combined signal, and its output(s) can be routed to the required external output(s). This also provides a master fader for the combined signal.*

## Understanding the Signal Path



# Standard DSP Mixer Plug-Ins

*Note that MadiXtreme cards do not have DSP mixing capability*

All mixer elements that can be inserted into a mixer strip are called 'plug-ins', including the standard elements which come with the SSL Mixer software, and optional plug-ins developed by SSL or third party companies. Any element can be inserted anywhere in the Mixer, before or after any other element. This section describes the standard mixer elements. Optional plug-ins are normally supplied with their own manual in electronic or printed form.

*NOTE: To change parameters of any mixer element or plug-in, it is necessary to be in Mixer Control mode. To switch from Mix Edit mode to Mixer Control mode, click on the Mix Control/Edit Toggle button in the lower right corner of the mixer window ('**edit**') or press **E**.*

## Plug-in Control Overview

Using the various mixer elements is straightforward. Some general guidelines are provided below which are valid for most mixer elements:

- Faders and sliders can be moved by clicking and dragging them with the mouse. Alternatively, clicking at the required position will cause the fader or slider to jump to that position. They are also controlled by positioning the mouse cursor over them and moving the scroll wheel or, if you want a finer control, by doing the same thing over the fader's value box.
- Pots (eg., the pan or balance pot for a send element) can be moved by clicking and dragging them with the mouse. Once a pot has been 'grabbed', the distance between mouse pointer and pot determines the resolution of the pot's response. Finer adjustments become possible as the distance is increased.
- Double-clicking on certain faders and pots opens dialogue boxes where the required value can be entered (eg., the send elements' faders, pan, or balance pots, and the fader elements).
- Values displayed in value boxes can generally be edited with the mouse buttons: right-clicking or left-clicking once in a value box respectively increases or decreases the corresponding value by one increment. Right-clicking or left-clicking in the selection box and holding the mouse button down increases or decreases the value continuously. The other mouse button can then be held down as well to speed up the process even more.
- Double-clicking in a value box often calls up a dialogue box where the required value can be entered using the computer keyboard.
- Buttons (bypass, mute, solo, etc.) respond to mouse-clicks. In some cases right-clicking them gives access to some extra functions (eg., the **MUTE** buttons in the multi EQ window).
- 'Nodes' in graphics can be clicked and dragged with the mouse, to shape an EQ curve in a multi EQ window or move a sound in a surround panner sound field window, for example.
- Standard Windows option boxes and check boxes are sometimes provided to select or activate/deactivate certain functions (eg., to select rectangular or triangular dithering and activate or deactivate noise shaping in the dither element dialogue box from the Audio Toolbox).
- Certain mixer elements have a name field in their main window and load/save buttons which open standard Windows load/save dialogue boxes.

## Sends

Several types of send elements are available: mono, stereo (with a balance control), mono to stereo (with a pan control), with or without Equal Power Panning.

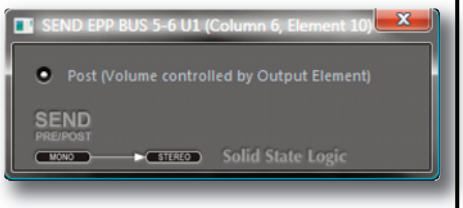
Any number of sends can be inserted anywhere in a mixer strip as required.

Any internal bus, external output or stream can be selected as the send destination using the I/O Assign tool (in Mix Edit mode).



All send elements have a mute button.

Double-clicking on a send element opens the following dialogue box:

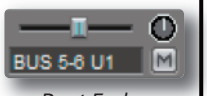



- Pre-fader sends (box unchecked) do not respond to changes made to output element settings, regardless of the output element's position in the mixer strip.
- Post-fader sends (box checked) respond to changes made to the fader, mute button and solo button settings of all outputs placed below them, except where these outputs are separated from the send element by an input element.

Please note the light blue identifying marks present on post-fade sends which allow you to see the pre/post status of a send at a glance.




*Pre Fader*




*Post Fader*

If you want to use a send element as an auxiliary send to an external effects unit, set the routing of that send element (using the Track Assign tool in Mix Edit mode) to the required external output(s). The wet signal can be returned via an input element within the same mixer strip (placed after the send to prevent feedback loops!), or to another mixer strip altogether.

If you want to use a send element as an auxiliary send to an internal plug-in effect, you can route the send to a bus, create a new mixer strip which has that bus as its input, and insert the required plug-in effect in the new mixer strip.

*NOTE: As well as sending a duplicate of the signal to the selected send destination, a send element still passes the original signal down to the next element below inside the mixer strip.*

## EQ 2-band

The 2-band EQs are fully parametric, allowing band-pass or notch filters with variable Q, level and centre frequency. You can add multiple EQs to a mixer strip to add more bands.

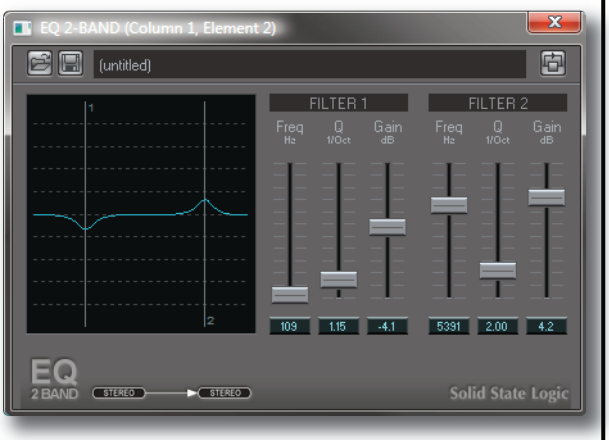



The Quality Factor (**Q**) determines how selective (narrow) the filter is. High Q means a narrow filter. The Q factor value is calculated by dividing the centre frequency with the bandwidth of the filter. If you are more used to working with 'octaves' instead of Q, '2 octaves' means that the bandwidth is twice the centre frequency and hence it has a Q of 0.5.

EQ parameters can be adjusted in real-time using the mouse buttons in the EQ parameters value boxes in the mixer strip, or using the scroll wheel. If you hold down both mouse buttons, the EQ parameters can be changed more quickly, and the direction of the change is determined by the mouse button pressed first. Double-clicking on a parameter opens a dialogue box where the value can be entered with the keyboard.

The mono 2-band EQ element has two separate fully parametric sections; For the stereo 2-band EQ, there are four linked active EQs, providing two bands for each channel.

Double click (with Mix Edit inactive) on the little Mixer Window to open the EQ-2 Band GUI.

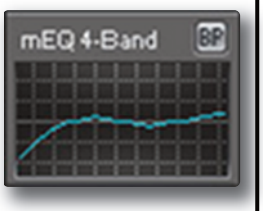



The response of the EQ is shown in the curve display and you can adjust the parameters using the faders or value boxes in the usual way.

Multiple EQ windows can be opened at the same time, the caption bar shows to which element in which column the GUI belongs to.

Below the caption bar the usual SSL Mixer GUI Top bar allows Loading and Saving of Presets and a Bypass button.

## Multi EQ




The multi EQ is a very flexible and DSP efficient Equaliser. Several variants of the multi EQ are available: 1-band, 2-band, 3-band, or 4-band, with each available in mono or stereo. As usual, more multi EQs can be inserted into each mixer strip if you need more bands.

The little Mixer Window shows the EQ-Curve right inside the Mixer strip. You cannot edit the EQ from here. Double-click the multi EQ element to open the main GUI, shown below.




If the signal overloads, the EQ curve will change from blue to red (also in the little Mixer Graph); If this happens, use the Input attenuation in the lower-left of the window to decrease the Input volume. (Click and drag the fader, click at the desired fader position, or use the value box.)

On the right of the input attenuator, there are tabs for each EQ band. Four parameter fields (**Type**, **Frequency**, **Q-factor**, and **Gain**) allow flexible manipulation of the currently selected band. Each band also has an individual mute button (**M**).

The **F-Type** parameter offers eight different options: low pass, high pass, band pass with constant skirt gain, band pass with constant peak gain, peaking EQ, notch filter, low shelving, and high shelving. Types are indicated by the icons on the buttons and by hover-over labels.

**Frequency**, **Q-factor**, and **Gain**, can be edited using the rotary knobs or the value boxes:

Knobs can be 'clicked and dragged' with the mouse. Once a knob has been 'grabbed', the distance between mouse pointer and knob determines the resolution of the knob's response. Finer adjustments are possible when the distance is increased. Alternatively, clicking at the desired position in the area around the knob sets the value immediately.

The values for the EQ bands' parameters can be edited by clicking (or clicking and holding) the small up and down buttons inside the value boxes. Alternatively, to enter a parameter value with the keyboard, simply double-click on a value box.

Left-clicking the mute (**M**) button toggles between mute and unmute of the EQ band. The button is highlighted when the EQ band is muted.

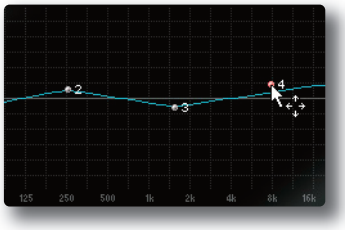
Right-clicking a mute (**M**) button 'solos' the corresponding EQ band, meaning that the other EQ bands become inactive/muted. When an EQ band is 'soloed', right-clicking **M** a second time returns all the buttons to their previous state, and left-clicking leaves all bands muted.

Right-clicking in the tab label will open a menu with options to restore default settings for that EQ band or for all EQ bands, to activate or deactivate solo mode for that EQ band.

The EQ curve can be edited directly in the graph display:

- The centre frequency for each EQ band is represented in the curve display by a 'node'. Each node has a number, so that they can all be identified easily even if they have been moved around a lot (any of the EQ bands can be set to any centre frequency, so the node for EQ Band 1 could end up being between the nodes for EQ bands 3 and 4). The nodes can be 'clicked and dragged' with the mouse. The frequency and gain parameters are adjusted accordingly in real-time.

**Dragging a node:**



**Dragging the line:**

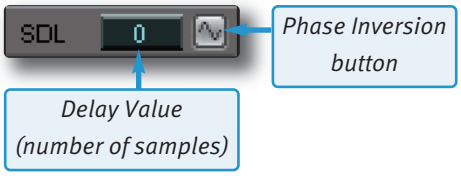


- The curve itself can also be shaped by clicking and dragging, to increase or decrease the bandwidth of an EQ band.

The Top bar of the Multi EQ contains the Standard Preset Load/Save and Bypass buttons.

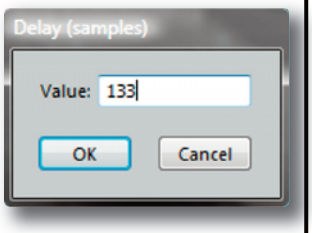
## Sample delay line

The sample delay line element can be used to compensate for time delays generated in the mixer or resulting from AD/DA conversion. It can also be used to invert the phase of a signal.




Signals can be delayed by up to 255 samples. The value can be edited directly in the value box by clicking the value and dragging the mouse, or by clicking the value and using the scroll wheel.

Alternatively, double-clicking inside the value box opens a text entry dialogue (shown right).




There is room for creative use of the sample delay line. For example, splitting a stereo pair of tracks into two mono tracks and inserting a sample delay line element in the signal path for one of these tracks can widen the perceived stereo image. Phase inverting the delayed signal can also add extra width, but may cause mono cancellation or phasing.

## Inputs and outputs

Mono, stereo, 4, 6, and 8 channel input elements can be inserted at any point in a mixer strip and can derive their signal from any external input(s) or internal bus(ses). All outputs have a mute (**M**) button.




*NOTE: An input always blocks any signal from mixer elements above. It can be used as an insert point at any position in a mixer strip, in combination with a send or output element (output followed by Input) it can be used as a send/return infrastructure. To remove it temporarily from the signal flow, please use the DSP MUTE tool.*

All the output types are also available inside the mixer strip:

- Outputs with fader: mono, mono to stereo with or without EPP (Equal Power Panning), stereo with or without EPP, 4, 6, or 8 channel, mono to LRCS, mono to LRC-LsRs
- Outputs without fader: mono, mono to stereo, stereo, 4, 6, or 8 channel

Input sources and output destinations can be changed with the I/O Assign tool, as described above.

*NOTE: As well as sending a duplicate of the signal to the selected destination, an output element also passes the original signal through to the next element below it in the Mixer strip where it is inserted.*

**Distinguishing between simple Input and Output Elements:**

'Output without fader' elements and input elements look very similar. There is always a little space above an output element and a little space below an input element to help identification.

## Faders

Mono and stereo faders can be inserted at any point inside a mixer strip. You can adjust the Gain between -96dB and + 6dB.

## Peak Meters

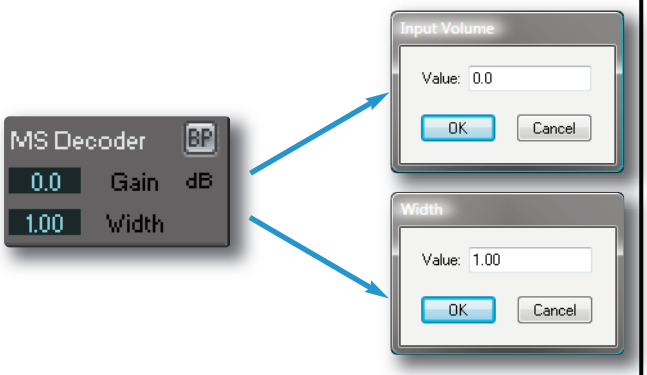
Mono and stereo peak meters can be inserted at any point inside a mixer strip.

*Mono peak meter* *Stereo peak meter*



## MS Decoder

The **gain** and **width** parameters for an MS signal can be adjusted directly in the mixer strip using the mouse buttons or scroll wheel. If you hold down both mouse buttons the values can be changed more quickly, and the direction of the change is determined by the mouse button pressed first.




Double-clicking a parameter value allows text entry with the keyboard.

Double-clicking on the mixer element opens the main MS Decoder window:

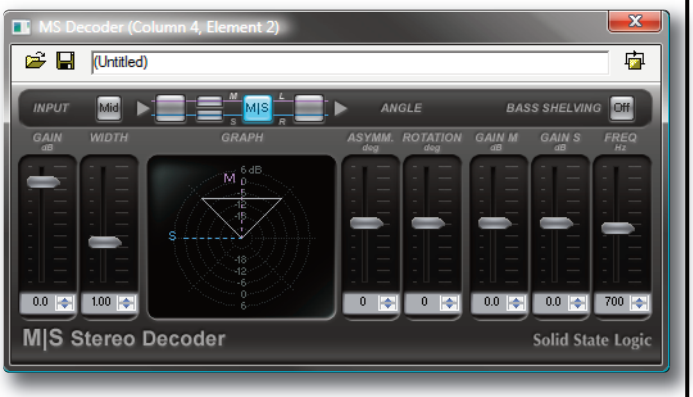
### Input Section

If the **Lock Mid button** is active, the relative gain of the mid MS matrix input signal is not affected by the 'width' setting.

If the button is inactivated, the relative gains of the mid and side MS matrix input signals vary at the same time and in opposite directions.

*Lock Mid Active* *Lock Mid Inactive*



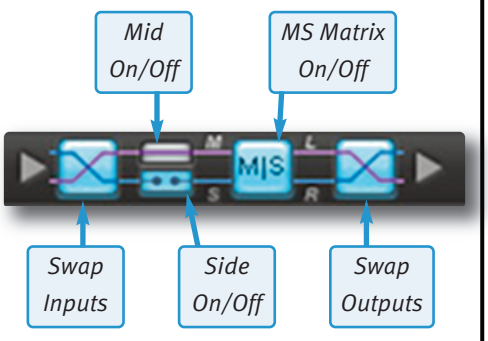

The **Gain** slider controls the overall input gain applied to both input channels. (-36.0 to 6.0dB)

The **Width** slider controls the ratio of the side MS matrix input signal gain to the gain of the mid MS matrix input signal. (0.00 to 3.00)




### Matrix/Routing Section

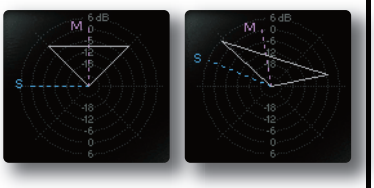
- The **Swap Inputs button** swaps the input channels before the signals enters the MS matrix .
- The **Swap Outputs button** swaps the output channels post MS matrix .
- The **Mid On/Off button** cuts the mid MS matrix input signal, so that the effect of the MS matrix on the side signal can be monitored easily.
- Clicking the **Side On/Off button** cuts the side MS matrix input signal, so that the effect of the MS matrix on the mid signal can be monitored easily.
- The **MS Matrix On/Off button** toggles MS decoding Matrix between on (illuminated) and off status.
  If MS decoding is turned off, the signals are still affected by the gain, width and bass shelving settings. However, the asymmetry and rotation parameters will have no effect on the output signals.



*NOTE: If the MS matrix is active, both input signals are attenuated by 6 dB. Then they are subtracted and summed to extract the left and right output signals.*

**MS Graph**

The graph display provides a visual representation of the perceived sound field, as defined and modified in real time by the parameter controls, and contributes to the intuitive operation of the MS Decoder.




The direction and length of the M and S vectors (dotted lines) represent the orientation and gain of the mid signal and side signal microphone respectively. The extent of the triangle (solid lines) in relation to the dB scale represents the overall gain; The angle between the M and S vectors varies according to the asymmetry parameter setting.

The overall orientation of the displayed sound field responds to adjustments of the rotation and asymmetry parameter settings.

**Angle Section**




- **Asymmetry** (–90, +90 degrees): The angle shift applied to the side MS matrix input signal along with the angle shift introduced due to the 'rotation' parameter. The value of the parameter has no effect if the MS matrix is not active.
  This parameter does not affect the mid MS matrix input signal at all.
- **Rotation** (–45, +45 degrees) Angle shift applied to both the side and mid MS matrix input signals. The angle shift of the side MS matrix input signal also depends on the setting of the asymmetry parameter.

**Bass Shelving section**




- **Gain M** (–6.0, +6.0 dB) is the bass shelving gain for the mid MS matrix input signal. This parameter is used in combination with the **Gain S** parameter to enhance the stereo perception at lower frequencies.
- **Gain S** (–6.0, +6.0 dB) is the bass Shelving Gain setting of the side MS matrix input signal. This parameter is used in combination with the **Gain M** parameter to enhance the stereo perception at lower frequencies.

  The Q-factor of both shelving filters is set at 0.7.
- **Frequency** (200, 1300 Hz) determines the cut off frequency for the mid and side bass shelving filters.

*Note: The MS Decoder can be used to check a stereo Arrangement for mono compatibility: Insert the MS Decoder into your Master Bus and set its parameter as follows: M/S Matrix: On to monitor in mono, Off to monitor in stereo (other top row buttons off), Gain: +6dB (0dB if monitoring via an EPP output element), Width: 1.00, Asymm: -90, Rotation: 0, Bass Shelving: Off).*

# Audio Toolbox plug-ins

***Important note:*** *The Audio Toolbox set of mixer plug-ins must be enabled by entering a password in the SSL Soundscape software (you can find the password in the product's registration card and on a sticker on the MX4 box). Entering the password for one of the Audio Toolbox plug-ins will automatically enter it for the rest of the plug-ins listed in the Options menu (Chorus/Flanger, Dynamics, Delay, Dither).*

The Audio Toolbox plug-in effects are a very useful collection of essential tools, bread and butter processes for everyday tasks. Chorus/Flanger, Dynamics processor, Delay and Dither elements are available for insertion in any mixer channel in all relevant mono, mono to stereo or stereo configurations.

The parameters shown in the mixer columns can be edited directly with the left and right mouse buttons or the scroll wheel. Double-clicking a parameter value field allows text entry. Double-clicking the name of an element in the Mixer channel opens the main plug-in window.

Clicking the **Open** and **Save** buttons in the top section of a main plug-in window opens standard Windows dialogue boxes which can be used to load or save presets.

There is also a bypass button in the top right corner of the window. The controls include faders which can be clicked and dragged, value fields which can be double-clicked to open dialogue boxes and where the right and left mouse buttons can be used to respectively increase or decrease the value.

## Chorus/Flanger




The Chorus/Flanger plug-in algorithm requires only a small amount of DSP resources. Mono, Mono to Stereo and Stereo versions are available. To produce the Chorus/Flanger effect, a proportion of the original signal is sent through a short delay. The resulting delayed signal is modulated by a low frequency oscillator (LFO).

In the main section of the window, the **Mix** fader controls the amount of signal that is delayed and sent to the LFO, the **Delay** fader controls the length of the delay, and the **FdBk** (feedback) fader controls the proportion of the processed signal that is sent back the input.




In the LFO section of the window, the **Freq**, **Phase** and **Depth** faders control the frequency, phase and depth (amount of modulation) of the LFO, the **Wave** buttons allow the selection of a sine or triangular LFO waveform shape, and the **Phase** indicator 'lights-up' to indicate the LFO speed and phase.

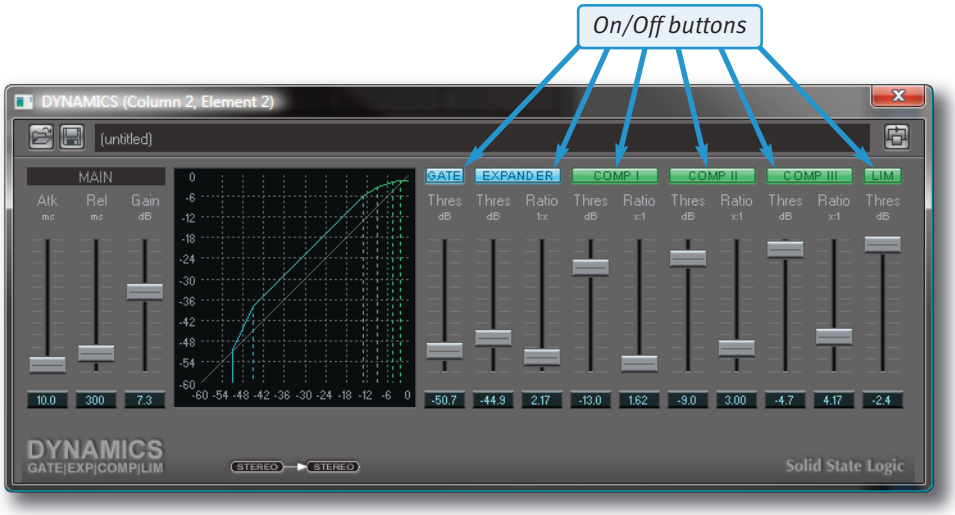
The Chorus/Flanger can produce anything from a slight thickening or doubling of the sound, to the common jangly effect that's great for guitars. Flanging is at the extreme end of the control range, with large pitch changes, delay variations and amazing stereo width and movement. It's great for Sound Design, alien voices etc.

*The parameter values can also be edited directly with the mouse in the mixer channel element.*

## Dynamics




The Dynamics algorithms are extremely efficient. Mono and Stereo versions are available. Up to 6 dynamic stages can be activated simultaneously. Each stage can be enabled or disabled in the main **DYNAMICS** window by clicking on the component labels above the faders to the right of the graphic. When on, Gate/Expander labels light blue, and compressor/Limiter labels light green.




**Dynamics Processing Components:**

- **Gate** - Allows real time gating of the signal to remove any low-level background noise between sounds and is useful for dialogue editing, guitar or vocal gating, or ducking backgrounds.
- **Expander** - This has similar operation to that of a noise gate, but has ratio setting which determines the slope of the cut-off. This means that, instead of total silence, the level can smoothly increase to allow a gentle build into the new sound. This can be used for higher levels of background noise removal, such as tape hiss or Air Con noise, auto-ducking, guitar hum etc or it can be used to 'pump the mix' which is commonly heard in dance music.
- **Comp I**, **II** and **III** (Three Node Compressor)- The main compressor section has three separate nodes, with each having its own threshold and compression ratio. This means that you can achieve a wide selection of effects from very soft (soft knee) to very hard (brick wall) compression. You can also have any combination of these, for instance, you could start with a soft compressor that leads into a much harder one. This compression section can be used for controlling the levels of dialogue, vocals, individual instruments like guitar and for maximising the energy in an entire mix.
- **Lim** (Limiter) - The limiter allows you to set a threshold for brick-wall limiting. With a zero attack time, the Limiter will not allow the signal to exceed the threshold. This is useful for such things as CD mastering or live PA, where it is vital that the signal never exceeds a given level; It is also useful for optimising broadcast levels during lay-back to tape.

All dynamics elements are also controlled via a **Main** section, where the attack time (**Atk**), release time (**Rel**) and output gain (**Gain**) are set. This allows you to define if the dynamics processor should track the input signal very quickly (short attack + short release) or smoothly follow the contours of the sound (long attack + long release). Finally, the overall gain section can compensate for the gain reduction caused by compressing a signal.

**Setting Attack and Release**

The attack and release controls determine how the output of the envelope detector responds to changes in the input signal and determines the how quickly the output gain is changed according to the Gain computer. Setting a long attack and release means that the control signal changes quite slowly and it will represent the average level of the input signal. With a compressor, this would result in an average level control, which increases when the average level decays.

Setting a fast attack means that the control signal reaches the peaks quickly, so that the applied gain responds quickly to signal peaks. This would be used for a fast limiter, that prevents the output signal exceeding a threshold. For a 'brick wall' limiter, the attack should be set to the minimum and the release should be increased until the signal sounds natural.

Setting a fast release means that the control signal responds quickly to a decay in the input level, so the gain can be increased immediately to compensate. The attack time should be increased until the signal sounds natural.

*NOTE: Setting both attack and release too fast will distort the signal, as the gain applied can change with the waveform of the input signal.*

The **Atk** and **Rel** faders have ranges of 0ms to 999.9ms for Attack, and 1ms to 9999ms for Release. The **Gain** fader allows the curve to be positioned in the available dynamic range, changing the output level by up to ±36dB.

**Setting Thresholds and Ratios**

There are 10 faders for setting thresholds and ratios for a Gate, an Expander, three separate Compressors and a Limiter. These allow very flexible control of the Compression Curve , which determines the transfer function between input and output levels.

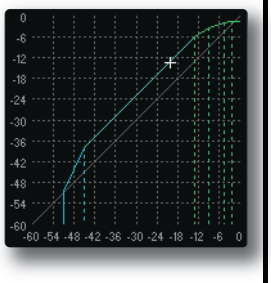
Depending on the number and type of processing stages activated, any curve shape can be defined, including 'soft knee' and 'hard knee' compressors and expanders, gates and limiters, or any combination of these.

The **Thres** (threshold) fader for each processor can be set from 0dB to -60dB and determines where the processor starts to kick in.

The **Ratio** fader for each processor sets the slope of the compressor curve (below the threshold for the Expander and above the threshold for each one of the 3 Compressors), from 1:1 to 1:50 (for expansion) or 50:1 (for compression), and determine the gain change to be applied at the measured control signal level.

*NOTE: The measurement of the input signal depends on the Attack and Release settings and higher values will slow down the animated gain indicator on the curve.*

The thresholds levels for the six processing stages need to increase from left to right. For example, the threshold for the gate can not be higher than the threshold for the limiter. Therefore, if a threshold fader is moved up and reaches a current position of another threshold fader to its right, this fader will also start moving up.




If a threshold fader is moved down and reaches the position of another threshold fader to its left, this fader will also start moving down. However, as long as the mouse button has not been released, returning the fader towards its original position will allow the other threshold settings to be restored.

Moving any threshold fader with the right mouse button will move all other threshold faders relatively, while maintaining the dB relationship. This allows the whole curve to be shifted up or down in the graph.

For the Gate and Expander, when the control signal is below the threshold the signal is muted, only above the threshold the 'Gate' is opened and audio passes through.

**Process Meter**

The amount of gain reduction is shown by the meter in the Dynamics mixer element. The meter calibration is 3dB per segment.

*No Processing* *Expansion/Gate* *Compression*

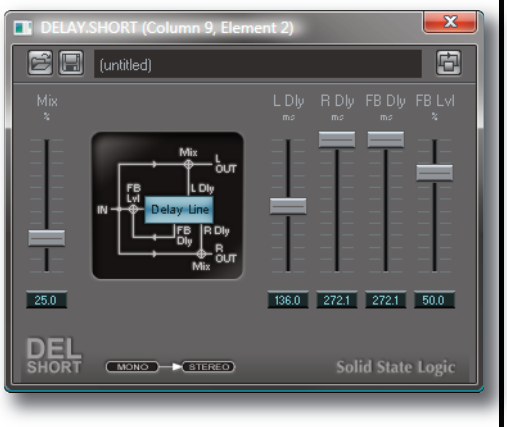



In the main **Dynamics** window, the level of the processed signal is shown in the curve display as a white '**+**' sign that moves in real time along the curve. This makes it easy to check whether the signal is operating at the correct point on the curve. The level of the input signal level is shown as a small vertical white line which moves along the horizontal axis .

## Delay




The Delay plug-in provides a 2-tap mono, 2-tap mono to stereo, or 2-tap stereo delay, with selectable cross-linking between left and right channels. Each can be a Long, Medium or Short delay.



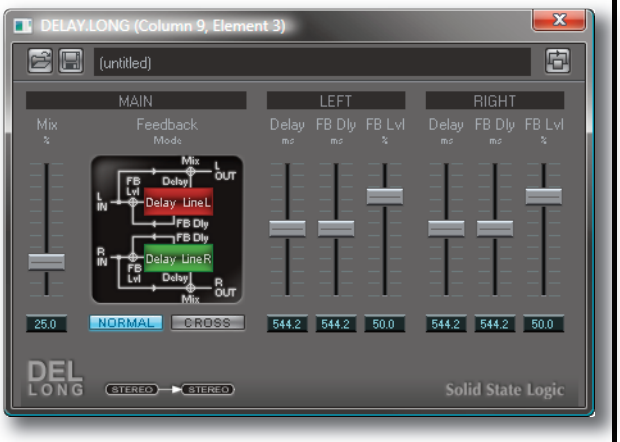

As longer delays need more DSP Memory, it is advisable to select the element type with the shortest delay time that suits your requirements.

For example, at 44.1kHz:

- Long: up to 1088.4ms
- Medium: up to 544.2ms
- Short: up to 272.1ms

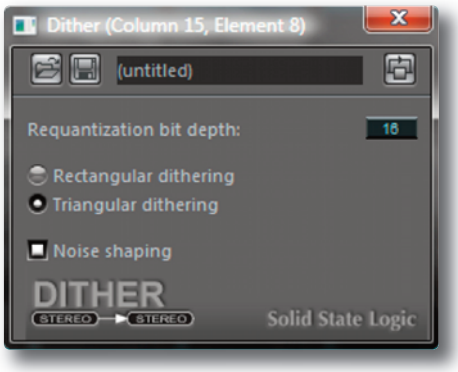
The **CROSS** or **NORMAL** button on a stereo delay defines whether the feedback portion of left and right delay are 'cross-coupled' for panning delay effects. When **CROSS** mode is active, the feedback signal from the left delay line is fed into the input of the right delay line and vice versa.

Stereo delay plug-ins feature completely independent controls for the Left and Right Delay Lines.




## Dither

The Dither element can be used when lowering the audio's bit resolution. It provides the ability to extend the dynamic range and include some of the low level detail of the audio signal that would otherwise be truncated and therefore lost. It provides **Rectangular** and **Triangular dithering**, with **Noise shaping** capability.




For instance, when multiple 24-bit tracks are mixed to stereo and the result is required for CD at 16-bits, the Dither element would normally be placed in the main stereo output channel strip. The recording resolution and the Dither Requantisation bit depth would then both be set to 16-bit.

The **Requantization bit depth** can be changed in the value box using the mouse buttons in the usual way. Alternatively, double-clicking in the value box will open a text entry dialogue.

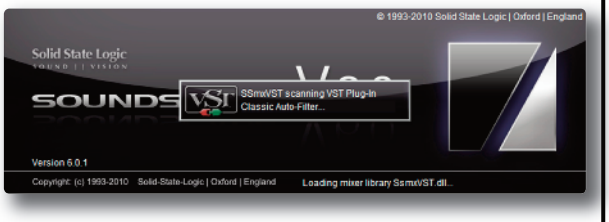
## Using VST Plug-Ins

VST effect plug-ins running on the host CPU can be inserted in the Soundscape Mixer just like any other mixer element. VST plug-ins are stored with a 'dll' extension. For a VST plug-in to be available to the SSL Mixer, the corresponding dll file must be loaded by Soundscape during startup.

By default, the Soundscape scans the 'C:\Program Files\Soundscape\MixElem\VST' folder and loads the dll files located in that folder. A different path can be specified by adding a 'VSTLibPath=' line to the file ssEditordef.ini file (located in C:\Soundscape), for instance: 'VSTLibPath=C:\Program Files\Steinberg\Vstplugins\'. You can open and save .INI files with Notepad.exe. Note that any subfolders of the designated folder will be scanned as well.

During scanning, a small window appears in front of the splashcreen where the plug-in names are displayed as they are loaded.

Dialogue boxes may also appear during that phase, for instance when a plug-in requires authorisation.

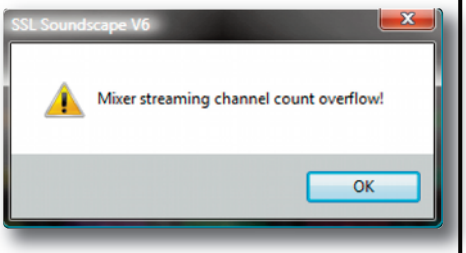



*NOTE: In rare cases VST plug-ins may not be loaded, because their 'unique ID' conflicts with the 'unique ID' of an already loaded SSL Mixer element. In such cases a warning message will be displayed.*

Since VST plug-ins run on the CPU of the host PC, the audio must be routed at high speed between the SSL Mixer and the PC in order to use VST effects at low latencies. This is done using the 'streams' provided by the MX4 or Mixpander. One stream is used per channel of audio, meaning a mono VST plug-in uses one stream and a stereo VST plug-in uses two streams. However, stream usage has been optimised so that consecutive VST plug-ins in the same mixer strip only require Input streams for the first VST Plug-In and Output Streams for the last VST Plug-in inside the chain. Therefore, as long as the signal does not leave 'PC Land' or 'DSP Land', no additional streams need to be set up to connect both worlds.

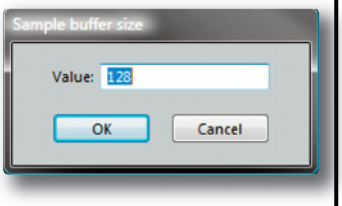
*NOTE: Track Elements are also inside 'PC Land', therefore a Track followed by a VST Plug-In only uses I/O streams once.*

Available streams are assigned automatically to VST plug-ins, taking into account that streams already used by a Windows application or assigned to a mixer input, output or send element are not available. If the number of available streams becomes insufficient for the number of VST plug-ins being used, the following warning will be displayed.




In this case the Mixer will also become inactive until the situation is resolved by deleting one or more VST plug-in(s), reassigning or muting one or more input, output or send element(s) or closing or re-routing a Windows application in order to release some streams.

The default buffer size for native mixer elements is 128 samples. This setting determines the 'latency' of the VST calculation. It can be changed in the **Settings**/ **Preferences** / **Mixer** / **Native mixer elements Sample Buffer Size** menu.




Typically, it should be raised if the PC is struggling to cope with the processing demands. Multiples of 64 up to 8192 can be used. If an invalid value is entered, the closest valid value below it will be used instead.

Operationally, VST plug-ins generally behave in a similar way to normal SSL Mixer elements; They can be moved or copied using the relevant tools, their settings are saved as part of the Mixer, and automation works in the same way as for other mixer elements.




However, the text box at the top of the main plug-in window behaves a bit differently; The arrow to its right can be used to open a program selection menu, and names can also be typed directly inside the box. The ***Enter*** key must be pressed to validate a newly entered name, otherwise it is lost as soon as a different program is selected .

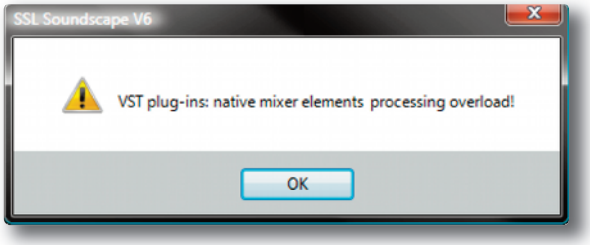
*NOTE: Many VST plug-ins have their own built in program/bank management facilities.*

## VST mixer elements

VST mixer elements are indicated by a plug-in name label with black text on a grey background.




If a processing overload occurs in the host PC, a warning message is displayed, the VST mixer elements turn red and the Mixer is deactivated:







Click **OK** to close the message box. To reactivate the Mixer, the configuration should be changed. This can be done quickly by selecting the Mute tool and muting or un-muting mixer elements.

## VSTi mixer elements

Everything mentioned above about VST plug-ins also applies to VSTi (VST instrument) plug-ins. In addition VSTi plug-ins have MIDI Port and Chan settings in the Plug-In Topbar to open a drop down list and select an available MIDI input port and MIDI channel.

*NOTE: In order to use VST instruments, a MIDI interface/Keyboard Controller must installed on the host computer and correctly configured.*

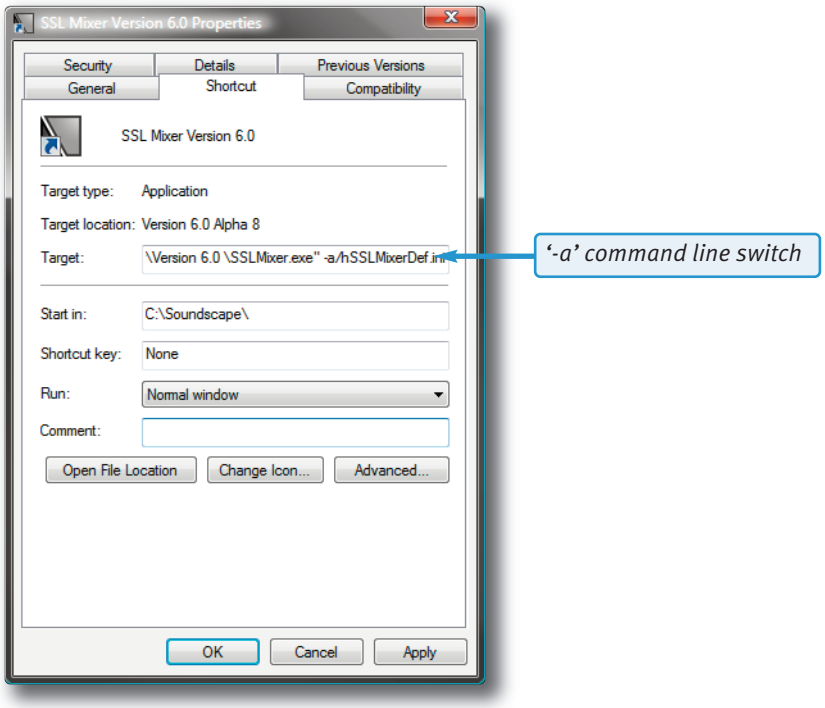
## Additional information regarding VST/VSTi plug-in compatibility

Certain VST or VSTi plug-ins require special Multi-Threading settings to work in the SSL Mixer environment. Typically when launching the SSL Mixer application 'problematic' plug-ins would cause an 'object initialization' error and the program would immediately close again. The '-a' command line switch solves the problem by initialising COM objects as 'APPARTMENT_THREADED' instead of 'MULTI_THREADED'.

To use the '-a' command switch, insert **-a** just after **SrEditor.exe** in the Target line of the Soundscape shortcut's properties.

In a standard installation, the SSL Soundscape V7.xx shortcut is located on your desktop. Right-clicking the shortcut will open the Properties window. The edited line should read as follows:

*C:\Program Files\Soundscape\Version 6.xx\Sreditor.exe" -a /h ssEditorDef.ini*




*NOTE: Using this command line switch may slow down the performance of the SSL Console Manager application.*

## Automatic Delay Compensation (ADC)

In a digital mixing system, any audio processing a certain amount of time, known as a processing delay. When several signals are processed in parallel (for instance a kick drum going through a compressor in a mixer column and a bass going through a compressor, an EQ and a chorus in an other mixer column), the cumulative delay for each signal may be different, resulting in 'misaligned' audio signals. Phase problems may also be created.

These problems can be solved by introducing compensating delays in some of the signal paths to get the output audio signals perfectly aligned. However, this introduces another problem: since all the signals must be aligned with the 'slowest' one (ie., the one that sustains the highest cumulated processing delay), any signal going through the system is delayed, either by a processing delay or by a compensating delay. A performer cannot, therefore, monitor their performance in real-time through that system.

A benefit of DSP-powered systems such as MX4 and Mixpander is that the processing delays are in most cases negligible, comparable to those of hardware digital mixers and effects units. In fact, MX4 and Mixpander are indeed hardware systems, controlled via a software interface.

On the other hand, native systems that use the CPU of the host PC are subject to higher processing delays inherent to the PC environment, which requires the use of buffers for reliable audio processing, even for the most basic effect plug-in.

VST compatibility offers SSL Mixer users a very wide choice of plug-ins. However, since VST plug-ins inserted in the SSL Mixer run on the CPU of the host PC, buffers must be used. The Automatic Delay Compensation takes care of the resulting alignment and phase problems.

Note that in recording situations, when real-time monitoring of the input signals is required, it may be preferable to use only DSP-powered plug-ins, with or without Automatic Delay Compensation. VST plug-ins can then be used when mixing, preferably with Automatic Delay Compensation active. The Automatic Delay Compensation (ADC) function can introduce compensating delays in the Mixer in order to counter the effects of any mixer element and bus delays. The ADC compensates for all bus and output delays. This means that the audio paths leading to common points are 'lined up' (ie., they present the same overall amount of delay). However, all buses and outputs are not necessarily lined up relative to each other.

For instance, the following output pairs are not lined up by default if they have completely isolated audio paths in the Mixer. They will only be lined up relative to each other if they are referenced by the same send or output element:

- MADI1-2 U1, used by the 'Output (Stereo)' element in mixer column 1
- MADI3-4 U1, used by the 'Output (Stereo)' element in mixer column 2

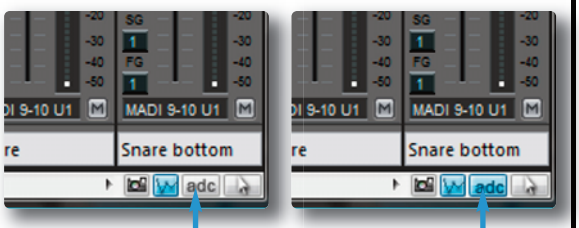
Automatic Delay Compensation toggle ***Ctrl+Alt+D***

### Automatic Delay Compensation toggle

The Automatic Delay Compensation can be activated or deactivated by clicking the **adc** toggle, located in the bottom right corner of the Mixer window.

Automatic Delay Compensation is active when the button is highlighted.

Alternatively, you can click **Automatic Delay Compensation toggle** in the **Edit** menu, or use the ***Ctrl+Alt+D*** key command.

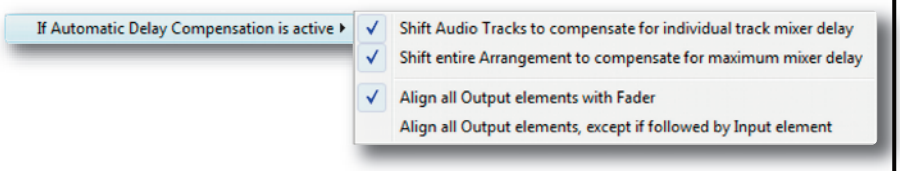



*Automatic Delay Compensation disabled*

*Automatic Delay Compensation enabled*

## ADC Preferences

Soundscape offers several options and strategies on how to compensate for processing delays.




The Options are centralised in the **Settings** / **Preferences** / **Mixer** / **If Automatic Delay Compensation is active** menu.

**Shift Audio Tracks to compensate for individual track mixer delay**

If this option is enabled, ADC shifts some of the audio Parts to the left (so that they will be played back early) where appropriate, before inserting delays in the Mixer.

This presents two advantages:

- The overall delay may be shorter, since only the delays caused by mixer elements placed below track inserts in the signal path need to be neutralised
- The ADC function will use less DSP and Memory resources to carry out its calculations

*NOTES:*
*The fact that certain Parts are played early is not reflected graphically by their position in the Arrangement.*
*The ADC is not 100% accurate while scrubbing*
*Audio playback may not start simultaneously for all audio tracks, due to the differing track delays.*
*The first samples placed at the beginning of the Arrangement for Audio Parts that are played early, are skipped.*
*This option is ignored when the Mixdown tool is used, which has its own strategy to compensate processing delays.*

**Shift entire Arrangement to compensate for maximum mixer delay**

If this option is enabled, the entire Arrangement is internally shifted 'to the left' (ie. to an earlier time position) to compensate for the maximum output delay in the mixer.

*NOTE: When this option is enabled, the first samples at the beginning of the arrangement are skipped. Therefore, when ADC is active and this option is enabled, it is best to not place any audio at the very beginning of the arrangement.*

**Align all Output elements with fader**

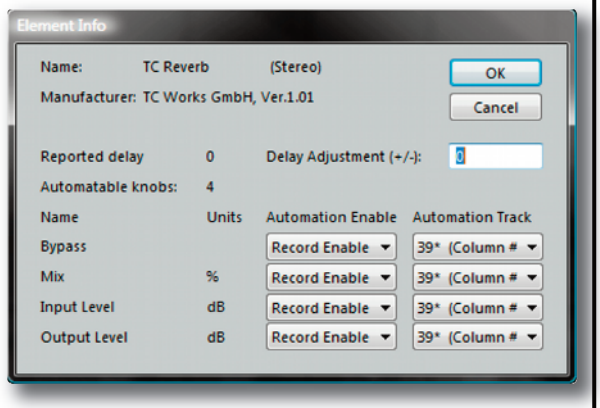
In order to line up outputs that have isolated audio paths in the Mixer, this option can be used. When selected, all output elements that have an Output fader are lined up. Outputs elements without a fader can be used for signals that should not be lined up (eg., a send or Column without fader feeding the artist monitor, Send/Return Path to an external FX processor, etc.).

**Align all Output elements, except if followed by Input element**

Enabling this option will line up all the output elements, except those that are followed by an Input element. This is especially useful, when the Output-Input Element Path is used as an Insert/Return for Outboard processors.

## Mixer element Delay Adjustment parameter

The ADC function relies on the delay values reported by each individual mixer element to calculate the compensating delays that must be applied. This value can be checked and edited by the user.




In **Mix Edit** mode, clicking a mixer element with the Info tool will open the corresponding **Element Info** window, as shown right for a TC Reverb plug-in.

The value in the **Delay Adjustment (+/-)** field will be added to or subtracted from the **Reported delay** value.

Any value (in samples) up to 99999 can be entered (more than should ever be needed).

For a negative value the minus sign must be entered. There is no need to type the plus sign for a positive value.

Note that this parameter is applied individually for each instance of a mixer element, allowing you to selectively correct or introduce a delay in the Mixer when ADC is active. The Delay Adjustment parameter setting is saved as part of the Mixer.

If you find that the reported delay value always needs to be adjusted by the same amount for each instance of a particular mixer element, you can override the reported delay value by adding a command line to that effect in the [MixWnd] section of the ssEditorDef.ini file (the path to the .INI file can be checked by right-click on **Soundscape Shortcut** / **Properties**, in the **Start in** field).

The ADC will ignore the value of the **Reported delay** parameter shown in the **Element Info** window and will use the value specified by the command line in its calculations, taking the **Delay Adjustment** parameter into account.

*NOTE: Soundscape must be restarted for the command line to take effect (settings specified in the .INI file are loaded on startup).*

The command line reads as follows:

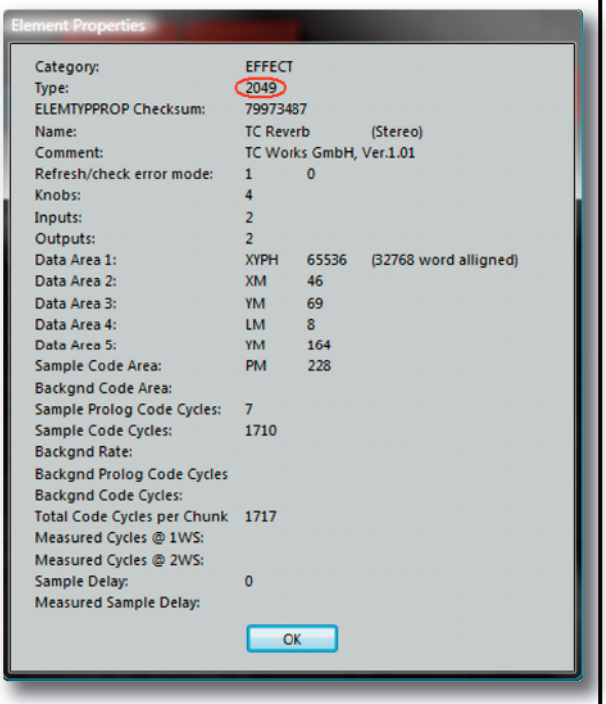
**ElemType XXXXXXXX Delay=Y // Element's name**

where **XXXXXXXX** is the unique Element Type ID for the element, and **Y** is the required delay value in samples. The 'Element's name' is only required for documentation purposes.

To check the Element Type ID, in **Mix Edit** mode, hold down the ***Shift*** key and click the mixer element with the Info tool. The **Element Properties** window for that mixer element will be displayed, as shown for the TC Reverb. The Element Type ID is displayed on the second line (red circle).

To continue with the same example, in order to specify a delay value of 45 for the TC Reverb stereo mixer element, the command line in the [MixWnd] section of the .INI file should read:

**ElemType 2049 Delay=45 // TC Reverb (Stereo)**



# Mixer Automation

The Automation reflects the general philosophy of Soundscape: it is flexible, powerful, easy to use and user configurable. Automation recording is in many ways similar to audio recording, and has been explained to some degree in the '*Recording Automation'* section of Chapter 3. A reminder of the main points, plus some information specific to automation recording which was not mentioned, will be provided in the '*Automation recording*' section of this chapter. 'Snapshot' automation recording will also be described.

Automation can be used easily with its default settings. However, these can be changed, so that for example you could record all the automation data for a whole Arrangement on one single automation track instead of using one automation track per mixer column. The '*Automation Setup window'* and *'Info tool*' sections of this chapter will provide all the information you need to customise the operation of the automation system to suit your preferences.

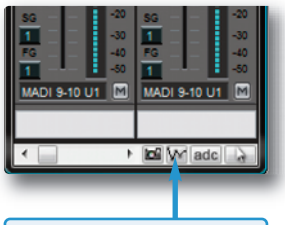
The '*Automation concepts and mixer controls behaviour*' section expands on certain basic concepts and features of the automation system and their practical implications.

Recorded automation data can be viewed and edited 'offline' with three specific tools which will be described in the '*Automation editing*' section.

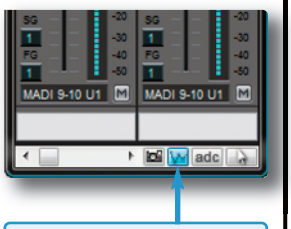
Some vocabulary specific to Automation will be used in this chapter. Automation data is made of 'automation events', and these come in two varieties: 'continuous events' and 'stepped events'. The difference will be explained in the '*Automation Event Editing tool*' section. 'Automation events' are generated whenever an automatable mixer 'knob' is used, and 'knob' is used as a generic term to cover all mixer controls, from faders and sliders to pots and buttons.

## Enabling Automation

For any automation data to be recorded or played back, the Automation must be enabled. This can be done by clicking the Automation Enable toggle which is located in the bottom, right corner of the Mixer window. Automation is active when the button is highlighted (blue):




*Enable Automation Status: Disabled*




*Enable Automation Status: Enabled*

Alternatively, you can press ***G*** on the computer keyboard to enable/disable Automation, but only when the Mixer window is open, or click **Automation Enable toggle** in the **Edit** menu.

## Automation recording

Automation can be recorded 'dynamically'. In this case, each time a knob is used, a corresponding 'automation event' is recorded in an automation Part. Automation can also be recorded 'statically' using the Snapshot function. In this case, the settings of the mixer elements at the time the Snapshot Automation Record button is clicked are recorded.

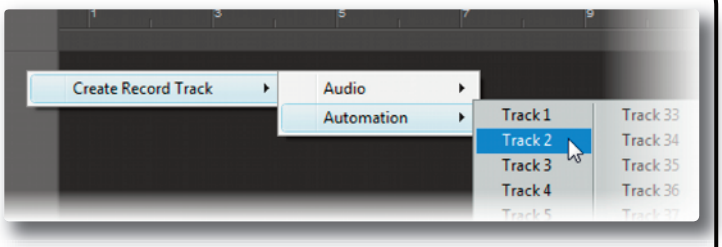
### Dynamic automation recording

Automation is recorded as Parts in the Arrangement which reference Takes on the disk, in much the same way as audio; 128 automation tracks are available, and each track can contain automation data for several mixer elements. By default, the automation data for all the elements in a mixer slot (including the mixer column itself and all the 'plug-in' mixer elements) is recorded on the automation track of which the number matches the mixer slot number.

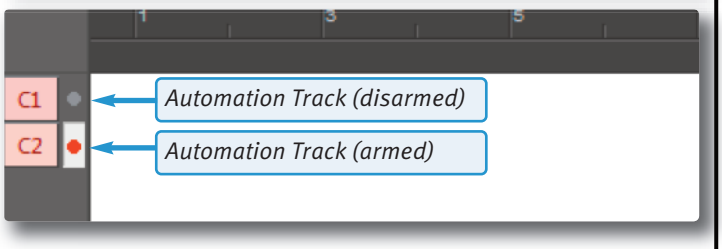



*Mixer Slot Number*

In order to record automation, an automation record track needs to be created by right-clicking in the Record Track Column:




It must then be armed using the track arming button, as described in detail in the '*Automation recording*' section of Chapter 3.

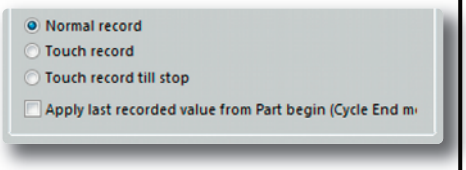



The procedures for audio recording apply to automation recording as well, with a few exceptions. For example, automation tracks cannot be armed from inside the Mixer or using the computer keyboard. However, loop recording, record loop stacks, manual punch in/out, and the Auto Punch In/Out mode can all be used for automation recording.

*WARNING: Unlike audio Parts, automation Parts can contain several layers of data (automation curves). Simply punching-in on an automation Part (in Normal Record mode) will overwrite all the data in the 'punched' section of this Part (ie., all the automation curves are affected). The 'Touch Record' mode or 'Touch Record Till Stop' mode should be selected for automation punch in/out with no risk of losing any data.*

## Automation Record Modes

There are three automation recording modes: **Normal Record**, **Touch Record**, and **Touch Record Till Stop**. These are selected in the **Automation Setup** window, accessed via the **Settings** / **Automation Setup** menu. The default setting is **Normal Record**.




### Normal Record

In this mode only the automation events generated during the current pass are recorded. If you are recording over an existing Part (or any section of a Part) on the same virtual track, this Part (or section of a Part) is overwritten. Overwriting a Part that contains several automation curves means that all the curves are overwritten regardless of which one is displayed, so great care should be taken when overwriting an existing automation Part. However, the Take referenced by the overwritten Part will be preserved on the disk. **Normal Record** would typically be used to record a new automation Part from scratch or when an existing Part needs to be replaced.

### Touch Record

In this mode, the automation events generated during the current pass are recorded, and all automation events from existing active Parts assigned to the same automation track are re-recorded to the new Part as well. Therefore, it is possible to overwrite an existing Part (or section of a Part) on the same virtual track, yet re-record all the data it contains at the same time as long as that existing Part is active when recording is started.

The knobs will respond to the automation data in existing active Parts. However you can grab them at any time to start generating and recording new data, with the mouse or using a control surface. When a knob is released, it starts responding again to the existing automation data, taking into account the settings of the **Knob auto return time** and **Knob touch timeout** parameters in the **Automation Setup** window (as described in the '*Automation Setup window*' section which is next in this chapter).

**Touch Record** would typically be used as a 'multiple punch in/out mode' for automation Parts (a punch in is performed each time a knob assigned to the automation track being recorded is grabbed, and a punch out is performed each time such a knob is released).

**Touch Record Till Stop**

In this mode, as in **Touch Record**, the automation events generated during the current pass are recorded, and all automation events from existing active Parts assigned to the same track are re-recorded to the new Part as well. However, unlike **Touch Record**, when a knob is released, it does not start responding again to existing automation data. Instead, this data is overwritten until recording is stopped.

**Touch Record Till Stop** would typically be used as a multiple punch-in mode for automation Parts, for cases when only the beginning of the existing automation curve(s) needs to be preserved.

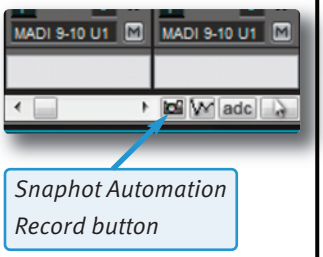
*NOTE: Everything that has been described here applies in the same way when several automation record tracks are used at the same time.*

**Cycle End Mode**

If the **Apply last recorded value from Part begin (Cycle End mode)** option is active (box checked), when recording is stopped, the last automation event recorded for each mixer knob during the last pass is shifted to the beginning of the automation Part, where it is placed immediately after the automation event representing the value for the knob at the beginning of that Part. All other automation events recorded during that pass are discarded.

## Snapshot automation recording

All the current settings for one or more mixer columns can also be recorded as a snapshot by clicking the Snapshot Automation Record button, located near the Automation Enable toggle in the bottom right corner of the Arrange window. Alternatively, press the ***F*** key or click **Snapshot Automation Record** under the **Edit** menu.




*Snaphot Automation Record button*

The resulting snapshot shows as an automation Part or group of Parts which starts at the time position where the Snapshot Automation Record button was clicked. One Part is generated for each armed automation record track, which contains information for the setting of each enabled automatable knob assigned to that track.

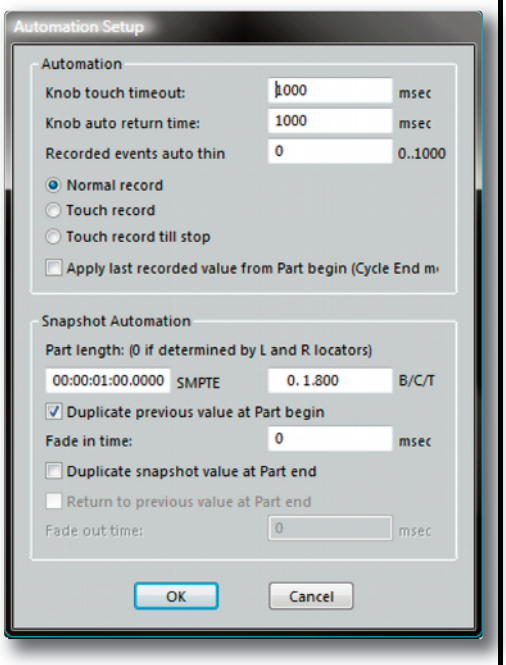
To record a snapshot for the whole mixer, one automation record track should first be created and armed for each mixer slot that contains a mixer column (unless the default settings are modified as described in the 'Info tool' section of this chapter).

The Snapshot Automation Record button can also be used during dynamic automation recording. In this case, the Parts will not start at the time position where the snapshot was made, (because that was by definition after the start of the recording), but all the snapshot data will be contained in the Parts at the time position where the snapshot was made.

The data recorded in a snapshot is also dependent on the settings chosen in the **Automation Setup** window, as described in the next section.

## Automation Setup window

Clicking **Automation Setup** under the **Settings** menu will open the **Automation Setup** window. In the top half of the window settings can be defined for dynamic automation recording and playback:




- The **Knob touch timeout** parameter is applicable in **Touch Record** or Play mode, when a control surface which is not touch-sensitive is used. In this case Soundscape receives no indication of the moment when an automatable knob is released, and should start responding again to existing automation data in a previously recorded active Part. Instead, Soundscape will assume a knob has been released if it is not moved for the duration entered in the **Knob touch timeout** box.
- The **Knob auto return time** parameter is applicable in **Touch Record** or Play mode. It determines the time that an automated knob will take to gradually return from its release value to the existing automation curve from an active automation Part with the corresponding track assignment.

*NOTE: The **Knob auto return time** and **Knob touch timeout** parameters apply only for knobs which generate continuous events (such as faders, sliders and pots). They do not apply for knobs that generate stepped events (such as buttons). Essentially, a button cannot move gradually from 'on' to 'off', and a button is not 'held' and then 'released', it just responds instantly to a mouse click (or a press of the corresponding control surface button). More information on stepped events and continuous events is provided in the 'Automation events editing' section of this chapter.*

- The **Recorded events auto thin factor** defines by how much newly recorded 'continuous events' will automatically be thinned, on a scale from 0 to 1000. This can also be done manually using the Automation Events Thinning tool, and it is advisable to try using the tool and getting a feel for the effect of the possible thin factors before deciding on an automatic setting, especially as thinning cannot be undone if it has been done automatically. Therefore, please refer to the 'Automation Events Thinning tool' section of this chapter for more information.
- The **Normal record**, **Touch record** or **Touch record till stop** option boxes and the **Apply last recorded value from Part begin (Cycle End mode)** checkbox are used to select the Automation Recording mode and enable or disable Cycle End mode. These modes have been described in the previous section of this chapter, '*Automation recording*'.

In the bottom half of the window, settings can be defined for snapshot automation recording:

- The **Part length** parameter determines the length of the automation Parts that will be generated whenever the Snapshot Automation Record button is used. Alternatively, entering '0' will cause the automation Parts to be generated between the Left and Right Locators, regardless of the current song position.
- When the **Duplicate previous value at Part begin** option is active (box checked), the last value for each curve in the previous snapshot or dynamically recorded automation Part (if there is one), or the 'intermediate value' (as described in the next section) which is current at the snapshot position will be duplicated at the beginning of the new snapshot, and followed by the value set for the snapshot. If no new value is set for the snapshot, the 'previous value' is still current and therefore is only recorded once.
- The **Fade in time** parameter determines how long it will take for automatable knobs that generate continuous events to move from the first (duplicated) value to the second (snapshot) value.

- When the **Duplicate snapshot value at Part end** option is active (box checked), the current values are recorded twice, at the beginning and end of the snapshot's automation Parts. This means that in effect automation curves of the chosen length are created, rather than 'single event curves'. Thereafter, these curves can be left active during Touch Record mode, so that any knobs that are moved eventually return to the snapshot value upon release. Using this setting, a basic setting for an Arrangement could be created as a snapshot of the same duration as the Arrangement.
- If the **Duplicate snapshot value at Part end** option is active, then the **Return to previous value at Part end** option is also available (otherwise it remains dimmed). If this option is selected (box checked), the last value for each curve in the previous snapshot or dynamically recorded automation Part (if there is one), or the 'intermediate value' (as described in the next section), which is current at the position where the snapshot Part ends will be duplicated at the end of the new snapshot, following the snapshot value. If the 'previous value' is the same as the snapshot value it is only recorded once. The 'Fade out time' determines how long it will take for automatable knobs that generate continuous events to move from the first (snapshot) value to the second (duplicated) value.

*NOTES:*
*The **Duplicate previous value at Part begin**, **Duplicate snapshot value at Part end**, and **Duplicate snapshot value at Part end** options only apply to automation curves relating to mixer knobs that generate continuous events. While the snapshot function can be used during dynamic automation recording, the settings described here will have no effect in this case.*

## Automation concepts and Mixer Control behaviour

### Snapshot vs dynamic automation recording

It should be noted that dynamic automation recording can only record data when changes are made to the settings of mixer knobs: automation data is composed of 'automation events', which are generated by using the mixer knobs and which can then be recorded in real-time (they can also be re-recorded when they are played back in **Touch Record** or **Touch Record Till Stop**). When no mixer knobs are activated, there is simply no data to record. This is why snapshot automation recording is necessary: it can create automation events out of the current settings of a static Mixer. Thereafter, the resulting automation Parts and automation events are no different from automation Parts and events that were recorded dynamically.

### Using a snapshot at the beginning of an Arrangement

When using Automation, it is generally advisable to record a snapshot at the beginning of the Arrangement, so that the initial settings are always restored automatically when the Current Locator is placed at the beginning. The snapshot could also be made the same length as the Arrangement, depending on requirements.

## Mixer Control response to automation data

### Automatic updating

The settings of all automatable mixer knobs are always updated automatically according to the position of the Current Locator and the automation data relating to that time position. This is not only the case when the Arrangement is played back. If the Current Locator is moved to a new position in the Arrange window by clicking at that position, or by entering a value in the Song Position readout or SMPTE readout, the mixer settings will be updated as well.

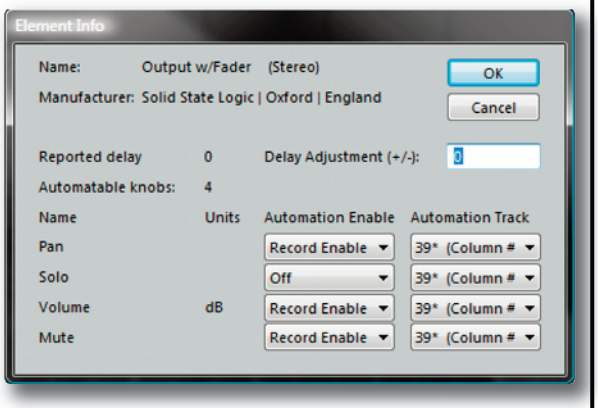
### 'Gliding' mixer controls and intermediate values

The settings of all automated mixer knobs that generate continuous events always change gradually from one value to the next, in such a way that if the Current Locator is moving between two automation events, the corresponding knob will 'glide' between the values of these events. For example, in the case of two distinct snapshot Parts which contain different values for the same volume fader, if the Current Locator is positioned between these snapshots, the volume fader will have an intermediate value between the snapshot values. In Play mode, this will result in a fade between the snapshot values. If such gradual changes are not required, the **Duplicate previous value at Part begin** box should be checked before the second snapshot is created, and any active automation Parts assigned to the same track and positioned after the intended position of the new snapshot should be muted while the snapshot is recorded.

Please note that due to this 'gliding' between values, in **Touch Record**, if an existing automation Part is left active while automation data is recorded to a new Part assigned to the same track at a time position after the existing Part (ie., to its right), then the mixer knobs which generate continuous events will still respond to relevant automation data in the existing Part during recording. They can be grabbed and moved to generate new automation data, yet as soon as they are released they return to the value determined by the existing automation Part, taking into account the **Knob touch timeout** and **Knob auto return time** settings. If in addition there is an active automation Part assigned to the same track after the position of the new recording, the knobs will also respond to the data in that Part: when released, they will therefore return to the intermediate value defined by the existing Parts before and after the recording position. If this is not required, **Normal Record** mode should be used (or the existing automation Parts should be muted before the new one is recorded). Note that the automation data in an active Part after the recording position has no effect on the mixer knobs during recording if there is not also an existing active Part before the recording position. Finally, in **Touch Record Till Stop**, by definition, upon release the mixer knobs do not respond to the data in active Parts positioned after the recording area.

## Using the Info tool on Mixer Elements

Automation events are generated when automate able knobs are used. All the standard mixer controls can be automated, including: mute and solo buttons, output volume faders, mono to stereo pan, stereo balance and surround pan sliders (X and Y, visible in the Surround Panner window), bypass buttons, all the EQ knobs for the 2-band EQ and multi EQ, send volume, and pan/balance pots. For send elements, even changing the pre/post status in the SEND dialogue box will generate corresponding automation data. The only exception is the track arming buttons on track insert elements, which obviously do not need automation! The situation is different with optional plug-ins. For example, while all the controls of the Audio Toolbox's Chorus/Flanger are automatable, only the bypass button, mix %, input and output levels of the TC-Works Reverb are automatable.



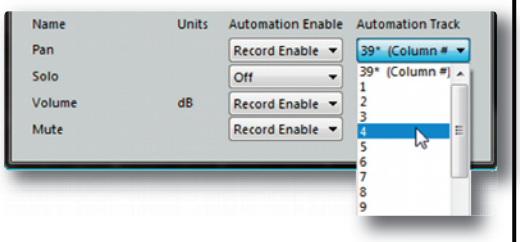

The Info tool allows you to check which knobs an be automated, to enable or disable automation recording and playback for each knob individually, and to assign each automatable knob to any automation Track. In Mix Edit mode, clicking on a mixer element with the Info tool will call up the **Element Info** window for that element.

For each automatable knob there is an entry for:

- **Automation Enable** selection box where you can select **Off** (no automation data will be recorded or played back), **Play Enable** (automation data will be played back but not recorded), or **Record Enable** (automation data will be recorded and played back).
- **Automation Track** selection box where any one of the 128 available automation tracks can be selected for the corresponding knob.

*NOTES:*
*Holding the **Ctrl** key while changing the setting for one parameter will apply the chosen setting to all parameters.*
*By default, all the automatable knobs are set to **Record Enable**, except the solo buttons which are set to **Off**.*

## Auto Renumbering Concept




By default the automation track number is the same as the mixer slot number. If you move the mixer column to a different mixer slot the automation track number will be changed accordingly and automatically for all the knobs in that mixer column. If you select a different automation track number in one of the **Automation Track** selection boxes, this automatic renumbering will not take place for the corresponding mixer element. If you want the automatic renumbering to take place, xx* (column #) must be selected, just as it is by default. This entry is at the top of the selection box's pop-up menu. for example, selecting '4' is NOT the same as selecting '4* (column #)', and the difference is that just selecting '4' will not let the automatic renumbering take place.

However, please note that if you move a mixer column to a different mixer slot and the automatic renumbering is enabled, any existing automation Part for that mixer column will need to be reassigned to a new track. This can be done by clicking on the Part in the Arrange window with the Info tool to open the **Part Info** window and changing the **Output** parameter, or by clicking on the Part with the Track Assign tool.

In most cases the default settings are perfectly adequate. You may want to change them for example to save space on-screen by recording automation for several mixer columns on one single track, if you need a lot of virtual tracks for audio Parts.

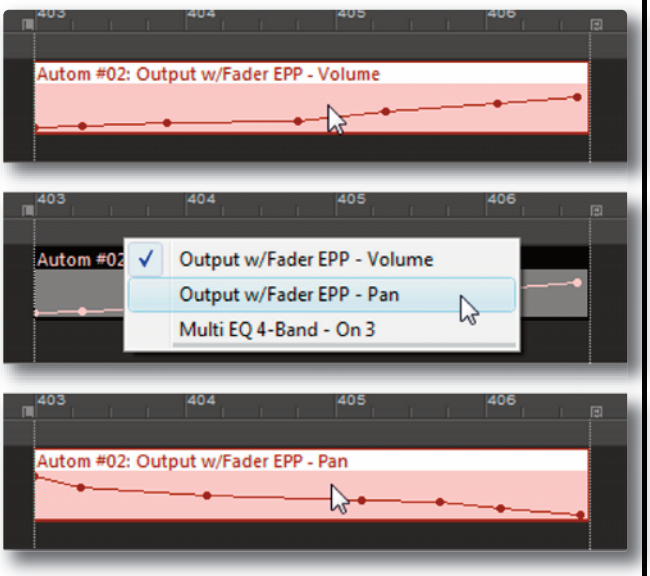
The Info tool can also be used in the Arrange window to check and edit information for automation Parts. In particular, it is possible to choose which automation curve will be displayed (this is also the function of the Automation Curve Select tool), and to reassign an automation Part to a different automation track for playback (for example in order to direct the data from that Part to different mixer elements, or perhaps to equivalent mixer elements in a different mixer column, all depending on current settings).

## Automation editing

Automation Parts can be treated in many respects just like Audio Parts. They can for example be moved, copied, cut, muted, assigned (to automation tracks), or deleted with the same editing tools as Audio Parts. However, 'Automation Events' within the Parts can also be viewed and edited with three specific tools: 'Automation Curve Select', 'Automation Events Thinning' and 'Automation Event Editing'. They have been mentioned briefly in the '*Editing Tools*' section of this manual, but will be examined in detail here:

### Automation Curve Select tool

An automation Part can contain many different layers of recorded data, generated by using the Mixer or adjusting plug-in parameters. This could include volume fader movements, mutes, reverb output level changes, etc. However, these automation Parts can only display one single layer of automation data, relating to one single knob, at any time. The 'layers' of data appear as 'automation curves', and the Automation Curve Select tool allows you to select which curve is displayed and therefore accessible for editing. Click on an automation Part with the 'Automation Curve Select' tool (***Ctrl+Shift+A***), and a menu will appear which lists all the automation curves contained in the Part. Hold the mouse button down, move the pointer to the desired entry in the menu and release the mouse button to switch to the chosen curve.




The Automation Curve Select tool can also be used on a group of selected Parts, but the menu will only show the entries relevant to the Part you click on, and any selected Parts that do not contain corresponding data will not respond.

*NOTE: The name of the currently displayed automation curve is automatically appended to the end of the Part name, which is visible if you have enabled* ***Show Part name in waveform*** *under the* ***Settings*** *menu.*

## Automation Event Editing tool

The Automation Event Editing tool (***Ctrl+Shift+E***) allows you to create, delete, or edit automation events. Automation events are shown as 'nodes' in an automation Part. Successive nodes are linked by a 'value line' and this graphically represents the 'automation curve'.

### Creating an automation event

Double-clicking in an automation Part with the Automation Event Editing tool will create a new event within the currently displayed curve. The new event's value will be determined by the vertical position of the mouse pointer, and its time position will be determined by the horizontal position of the mouse pointer. If the snap setting is active, the event will be created at the nearest snap point. The value line will automatically link the new event node to the nearest previous and/or next existing event nodes.

*NOTE: If needed, use the Automation Curve Select tool first to display the required curve. Event creation is only possible if at least one event of the same type has already been recorded in the Part (ie., if there is already an automation curve).*

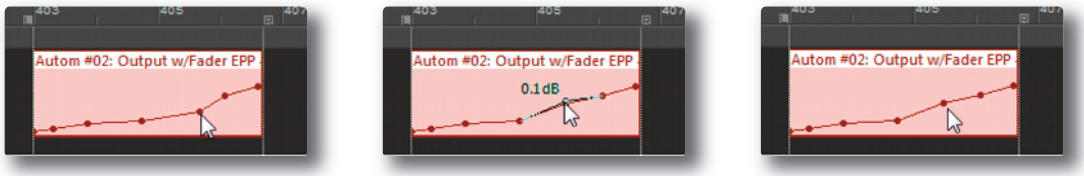
### Deleting an automation event

Double-clicking on an existing event node will delete it. This needs some precision and zooming in to an appropriate level will sometimes be helpful (when a lot of events are displayed in a small space on the screen).

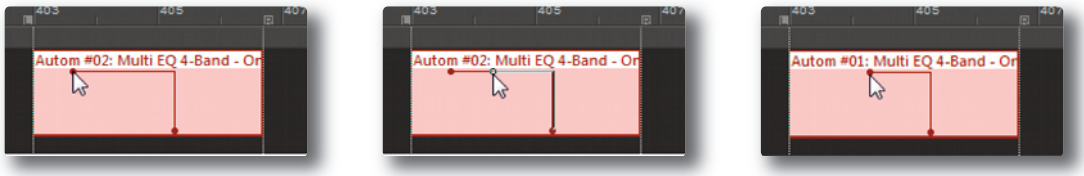
### Editing an automation event

Clicking on an event node and dragging it vertically will change its value, while dragging it horizontally will change its time position. If the snap setting is active the event node will 'jump' to the nearest snap point when the mouse button is released.

For automation 'continuous events' (the kind of events created by eg. a volume fader or a pan or EQ gain pot, which generate a continuous stream of events with many possible values), the target value will be displayed and updated in real time as you drag the event node, as in the second screenshot below, until you release the mouse button:




For automation 'stepped events' (the kind of events created by a solo, mute, bypass, or phase inversion button, or a group of EQ curve selection switches, all of which can be in one of a limited number of positions), no value is displayed as you drag the event node, and you can only move it to 'allowed' locations (for example for a multi-EQ 'on' button the node can only be at the top of the Part (button 'on') or at the bottom (button 'off').




*NOTE: The edits carried out with the Automation Event Editing tool result in a new Take being created. In that sense the Automation Event Editing tool acts as a processing tool: each time an event is edited, the corresponding Take is processed. Therefore, this tool is subject to the setting chosen under* ***Settings*** */* ***Preferences*** */* ***Arrangement*** */* ***Automation Processing tools behaviour*** */* ***Automation Takes that become unused after Processing****: if* ***Always delete*** *is selected, or if* ***Ask to delete*** *is selected and you do delete unused Takes, this may restrict your use of the 'Undo' function, especially as if an event is deleted and then another one is created, the newly created event will take the place of the deleted one on the disk.*

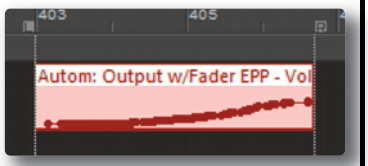
## Automation Events Thinning tool

This tool is also accessed via the ***Ctrl+Shift+F*** keys. Automation 'stepped events' as defined in the previous section do not require thinning, because they are generated one by one exactly as required. Therefore, by design, the Automation Events Thinning tool has no effect on them. Automation 'continuous events' on the other hand, can be generated in enormous quantities. For example, if you move a fader with the mouse, a new continuous event is recorded each time the fader moves by one pixel. If a MIDI controller is used via the Console Manager, a new event is recorded for each MIDI controller message that is received by Soundscape.

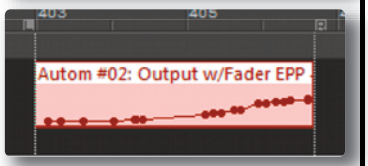
Inevitably, some of these automation continuous events are redundant and in some cases undesirable. For example, the best way to define a perfectly smooth linear volume fade, or a perfectly smooth left to right panning change, is to have very few events linked by a straight value line, as opposed to a great number of events linked by as many sections of the value line, each of these sections with a different slope and length.

'Thinning' is the process of eliminating redundant or otherwise unwanted automation continuous events. In Soundscape, 'thinning' is the result of extremely thorough programming and is carried out with discrimination. The first and last events of an automation curve, but also any important 'turning point' events (such as the event at the beginning of a change of 'rate' within a fade), are preserved even at the highest 'thin factor'.

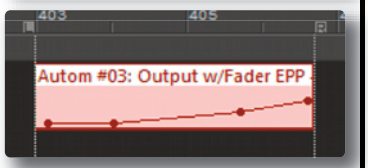
This example shows the recorded automation curve for a volume fader movement performed with the mouse:




This shows the same volume fader automation curve, after processing with a thin factor of 200:



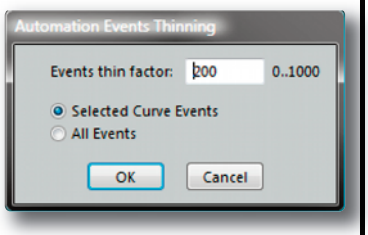

Here, the maximum thin factor of 1000 has been applied to the original automation curve:




Like any other editing operation in Soundscape, thinning can be undone if the result is not as expected (subject to the restriction mentioned in the note below). You could also make multiple copies of the relevant automation Part, process them with different 'thin factors', and use the Track Assign tool to play them back one by one before making your choice.

*NOTE: The Automation Events Thinning tool is a processing tool. Using it results in a new Take being created. Therefore it is subject to the setting chosen under* ***Settings / Preferences / Arrangement / Automation Processing tools behaviour / Automation Takes that become unused after Processing****: if* ***Always delete*** *is selected, or if* ***Ask to delete*** *is chosen and you delete unused Takes, this may restrict your use of the Undo function, especially as, if an event is deleted and another one is created, the new event will take the place of the deleted one on the disk.*

Click on an automation Part (or group of selected Parts) with the Automation Events Thinning tool. The dialogue box shown right will appear. Enter the thinning factor (any value between 0 and 1000). If only the automation continuous events currently displayed in the Part should be changed, choose **Selected Curve Events**. If you want to thin all the automation continuous events contained in the Part, choose **All Events**




*NOTE: Thinning can also be carried out automatically after automation recording.* ***Recorded events auto thin factor*** *within* ***Automation Setup*** *automatically applies a thinning factor to any new automation recording.*

## Mixer editing with existing Automation

The following information is vital if you intend to edit your Mixer when some automation Parts have already been recorded.

When automation data for several mixer elements of the same type (eg., several stereo sends) is recorded to the same automation track, then it is an element's 'index number' that will determine the connection between the data and the element. For example, if you record automation data for the second stereo send element in the mixer, this data is stored with a 'stereo send #2' reference. Consequently, on playback, this automation data will be directed from the track to the second stereo send element in the Mixer (counting from top to bottom within mixer columns and from left to right across the mixer) that is assigned to that automation track.

Therefore, if a new stereo send element is inserted (or copied) before an already existing one and is assigned to the same automation track, any already recorded automation data will end up reaching the wrong mixer element, because it refers to the position of a stereo send in an order that has been modified. Similarly, if a mixer element is deleted when several elements of the same type are present after it in the mixer (below it in the same mixer column or in a mixer column to its right), and if these elements share the same automation track assignment, they will all move up by one index number, and respond to the wrong automation data.

Moving a mixer element to a position before or after another element of the same type and assigned to the same automation track means that the index numbers for these elements are 'swapped'. They will both respond to the wrong automation data. If more than two elements are 'reordered' in this way, the result will be even more confusing.

Finally, moving a mixer column to a different mixer slot will cause any existing automation data to miss its target for the mixer elements in that mixer column that have the automatic renumbering function enabled (as described above in the '*Info tool*' section of this chapter). In such cases the Part(s) containing the automation data will need to be reassigned to a new track, whose number matches the new mixer slot number.

### Resume

- When a new mixer element is added to the Mixer after automation recording has taken place, it is advisable to assign all its knobs to an unused automation track using the Info tool, unless you are certain that no other element of the same type is already assigned to the same automation track and has automation data recorded for it.
- While deleting a mixer element will change the relative position, and therefore the index number of all other elements of the same type placed after it in the Mixer, muting an element will not. In fact, a muted element will still respond to automation (its knobs will move), even though it appears dimmed. So if you are in any doubt as to what automation data has been recorded and on which track, muting an element is safer than deleting it.
- Moving an element to a new position relative to another element of the same type that has the same track assignment is best avoided. It is better to mute the element, create a new one at the chosen position, and assign that new element to an unused automation track. It may be necessary to reassign existing automation data for the element which has been muted to the track selected for the new element. This cannot be done directly if this data is stored in a Part which also contains other automation curves. Instead, the required data will first need to be extracted to a new track using the procedure described on the following page:

### Extracting an automation curve from a Part containing several curves

- Using the Info tool, disable automation for all mixer elements assigned to the same automation track as the Part containing the automation curve you want to extract, except for the mixer element that this automation curve is directed to. For this mixer element automation must be enabled.
- In the **Automation Setup** window, select **Touch Record** mode.
- Create a new instance of the automation track that the Part containing the targeted automation curve is assigned to in a new virtual track slot of the Record Track Column (choose an empty virtual track to avoid overwriting the original Part).
- Make sure that the Part which contains the targeted data is active, and start recording.
- The resulting new automation Part will contain the targeted automation curve and nothing else. You can now assign it freely to any automation track, or copy it, etc.

# Console Control

## Introduction

The Soundscape console control interface offers a deep level of remote control from MCU-capable SSL Control Surface and other MCU-capable controllers by 3rd parties (results may vary). It is based on multi-bay MIDI communication. SSL Consoles provide the option of MIDI over Ethernet, which is fast, reliable and cost effective. Console control also works with MIDI Interfaces, virtual MIDI devices (ie. iPad MIDI over WLAN) and inter-application "midi cables" (ie. LoopBE).

SSL console control is in addition to the Legacy Console Manager which is still available for those who would prefer to use it – see the *Legacy Console Manager* section of this chapter for more details.

In order to use Console Control with an SSL Control Surface, you will need to select the Soundscape Default Profile on the Remote Browser associated with the Control Surface, as described in the user documentation which came with your SSL product (also downloadable from the SSL website: www.solidstatelogic.com). If no Soundscape profile is available, you may want to start with the Logic Default Profile.

We strongly recommend making sure that the Remote Browser software is up to date.

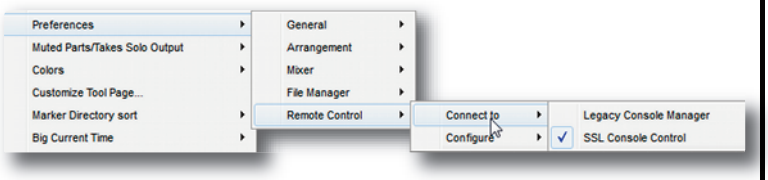
*Note that the following descriptions use SSL Nucleus and Matrix as examples, though most functionality also applies to other SSL consoles or 3rd Party MCU compatible controllers (results may vary).*

## Feature Overview

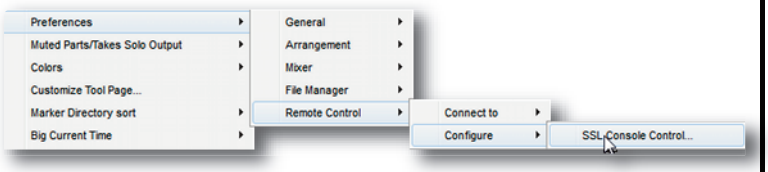
- Support for a maximum of 6 MCU controllers (48 Faders)
- Plug In control from channel V-Pots with defined plug in control bays independent of total number of control bays in use
- Plug In Name Display
- Access to surround panning parameters from V-Pots
- Aux send control including Mute and Pre/Post Switching
- Channel View (vertical) and Multi Channel View (Horizontal) V-Pot control of sends and pan parameters via cursor key paging
- Fader/V-Pot parameter Flip or Duplicate eg. EQ gain control from faders
- Name/Value option for V-Pot displays
- SMPTE/Bar Beat timeline display option (if Console has a TC Display)
- Selected (SEL Key) channel strip display on Soundscape Mixer Screen
- Active Bay(s) display on Soundscape Mixer Screen
- Set Routing, Solo- and Fader Groups from mixer front panel with on-screen list view
- Direct SSL Console Bundle plug-in Access Keys (EQ/Dyn)
- Full support for the four 'modifier' keys for 'Option All' type commands
- Channel Metering
- Track Arm Recording and Automation Tracks
- Control of Automation and Snapshot Master Modes
- Dedicated Automation and Snap Enable commands
- Set L R locators via Jog Wheel
- Next/Previous Marker locate
- Open/Close Plug In Windows on-screen while editing them
- Open Close Routing Lists on-screen while editing them
- Pre-assign Edit commands to 16 MCU function keys, and use the modifier keys to assign to left /right/***Alt*** mouse clicks
- MTC Input: read and follow MTC (Input) using IP MIDI as well as standard MIDI Drivers
- MTC Output: generate MTC (Output) using IP MIDI as well as standard MIDI Drivers
- MMC Output: generate MMC (Output) continuous locate commands from Play Pointer Position using IP MIDI as well as standard MIDI Drivers (WDM and MME)

# Activating and Configuring SSL Console Control

Once SSL Console Control has been installed, it is activated within Soundscape by selecting **Settings** / **Preferences** / **Remote Control** / **Connect to** / **SSL Console Control**

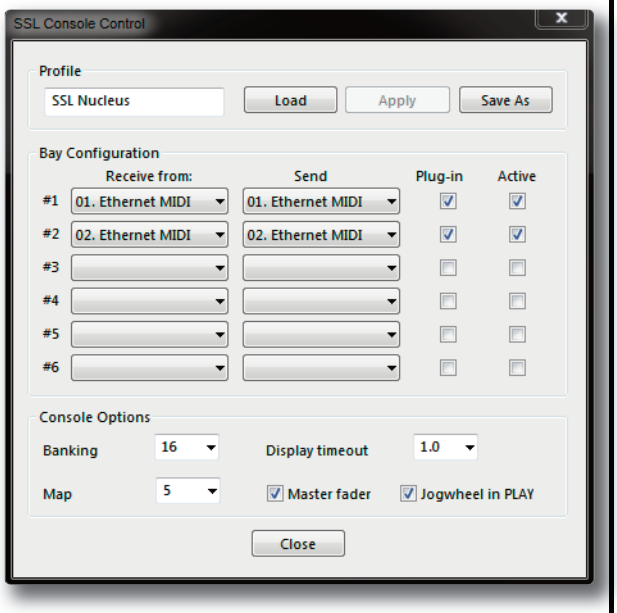



**Settings** / **Preferences** / **Remote Control** / **Configure** / **SSL Console Control...** opens the **SSL Console Control** configuration window, shown below.




*NOTE: SSL Console Control only supports MIDI Devices that are set to MCU (Mackie Control Universal) Control Protocol*

At the top of the configuration window, you can give the configuration a profile name. There are also **Load** and **Save As** functions which allow you to load and save profiles via standard Windows navigation dialogues.




## Bay Configuration

Up to six 8-channel bays can be assigned to Soundscape control. For each bay, select the MIDI channels used for sending and receiving control data for that bay.

The **Plug-in** check box allows you to choose whether or not the rotary encoder section shows Plug-In Parameter Pages (primarily for use with SSL Duality, to keep controls near the centre section).

You will also need to activate the bay by checking the **Active** box.

## Console Options

The following three values can be defined by clicking the down arrow to the right of the number:

- Display The **Banking** number determines amount of channels scrolled when console's Bank <–> buttons are used;
- Display **Map** defines the Soundscape Tool Page Number that is used to determine the 16 Tools selectable with the F1-F16 controls;
- **Display Timeout** defines the time the value labels on the console take to revert to their name label

If you select the **Master Fader** check box, the first column named **MASTER** in Soundscape's Mixer will be mapped to the console's Master Fader. Selecting **Jogwheel in PLAY** allows the transport to respond to Jog and Scrub wheel movements while Soundscape is in Play.

*NOTE: As with all Soundscape settings, changes are only recalled on next startup unless a **Settings** / **Save Settings** command has been applied. However due to the fact that SSL Console Control does save profiles on its own, please make sure to save the profile changes and use **Save Settings** when you are happy with a change.*

## V-Pot Operation

Channel V-Pots and associated V-Sel switches are used to control the main mixer parameters, with the exception of Channel levels (which are controlled with the motorised faders, except when FLIP is engaged) and the channel Solo and Cut functions which have dedicated front panel keys. Mixer functions are assigned to the V-Pots via four Parameter Group select keys: **SENDS**, **PAN**, **TRACK** and **PLUG -IN**.

*Please refer to the DAW appendix of the relevant SSL console's Operator's Manual for details of how Soundscape control appears on the control surface.*

2 Distinct View Modes:

- Channel Strip View where the eight V-Pots in a selected channel's associated bay, control a pre-defined parameter set for the selected channel
- Multi Channel View, where each V-Pot is controlling the same parameter across the individual channels. Parameter names and/or values are displayed on the lower display row, leaving the channel labels on the top row

The Channel SEL key toggles channel selection in Channel Strip View. Normally only one channel can be selected at any one time, however in combination with the modifier keys, multiple channels can be selected for mode selection and routing operations. The selected channel can be cleared by a second operation of the SEL key. An on-screen indication of the selected channel is shown inside Soundscape's Mixer.

Switching between the two view modes:

On initial selection all Parameter Group switch defaults to Multi Channel View. Selecting the currently active parameter group switch a second time flips to Channel Strip View.

A temporary Mode View, available for choosing additional control and display options by holding down a parameter group select switch, is available on the lower V-Pot displays and actioned via the V-Sel switches in an eight channel bay. The displays are repeated in each controller bay.

Two additional Keys, DYN and EQ override the current Parameter Group selection and directly assign the SSL EQ or Dynamics Plug in parameters on the currently Selected channel to V-Pots in Plug In Channel Strip View.

If no SSL plug-in is assigned to the selected channel, then the EQ and DYN keys have no action.

There are two further keys associated with V-Pot functions, NAME/VALUE, which toggles between displaying parameter names and parameter values and replaces the channel labels with additional information in Multi Channel View, and FLIP which assigns the current V-Pot parameters to the faders.

In NAME mode, when a V-Pot is operated, the lower V-Pot label displays the parameter value as the control is turned, reverting to the parameter Label after a defined time out period.

System defaults to PAN in Multi Channel View with the NAME display on the V-Pots when first initialised.

## Parameter Group Details

### Sends

Selects the mixer strip sends in Soundscape's Mixer for V-Pot Control

V-Pots default to Multi Channel View, controlling the first Send (Send 1) level. The V-POT label shows Send 1 on all channels with a valid send plug in. On channels with no send of that number dashes (----) are displayed.

NAME/VALUE show level in dB (one decimal point is fine ie. –10.1 dB) as Value.

The indicators around the V-Pot show send level as a continuous ring (wrap mode) with off (∞) as the 1st (leftmost) segment. Unity gain (0dB) is indicated by the red led beneath the V-Pot illuminating.

V-Sel controls the send mute/on status. Send Mute is indicated by all segments off, the lower red led is switched on.

The Up/Down console cursor keys increment or decrement the send no. being controlled. Send number is indicated in the consoles 2 Digit Mode Display.

The left/right cursor keys flip the V-Pot to controlling Send Pan (on stereo sends) followed by Pre/Post send source selection on the V-Sel Switch. The V-Pot is not active when Pre/Post is selected and V-Sel is not active when Pan is active.

In Send Pan Mode, the V-Pot controls pan position, indicated by single dot around the encoder. When the pan pot is center, the lower red led is illuminated.

For Pre/Post selection the Lower V-Pot label display the current source, Pre or Post, which can be flipped via the V-Sel switch.

The Left/Right cursors keys wrap around between the three options. Level – Pan – Pre/Post, ie. in Pre/Post the Right cursor reselects Level, or in Level, the Left cursor selects Pre/Post.

FLIP assigns the current slot Send levels to the faders, the Sends mutes to the channel cut keys, the Send pan to the V-Pots, and Pre/Post source selection to the V-Sel Key.

Selecting Channel Strip View (by pressing the SENDS Parameter Group button again) operates only on the V-Pots in the 8-channel fader bay containing the currently selected Channel.

Once active, Sends Levels 1- 8 are assigned to the V-Pots.

If no send has been assigned to a slot in the selected mixer strip ' ------ ' is displayed in the empty send slots.

NAME/VALUE shows send level in dB instead of Send number, and the V-Sel keys control the send mute function. If more than 8 slots are in use, the Left/Right cursor keys act as send slot page keys.

FLIP assigns the eight sends level on the current send page to the channel faders, the send mutes to the channel cut keys, the send pan to the V-Pots, and pre/post source selection to the V-Sel Switch.

Use of the Modifier Keys to globally or partially toggle switch functions ie. send pre/post source selection

***OPT+*** V-Sel SWITCH toggles function on all channels

***SHIFT+OPT+*** V-Sel SWITCH toggles function on selected channels (Via the SEL key)

**Temporary SENDS Mode View**

Holding the SEND key displays the Mode View in the V-Pot Display. Via the Send Mode View the user can set the busses, outputs or stream destinations for the mixer sends.

'Route' will be displayed beneath the V-Pot 1 in each bay. Selecting Route via the V-Sel 1 key will select Routing Mode, pressing any V-Sel of any channel again will display the current bus/output/stream or bus/output/stream pair assignment for the sends in either Strip View or Multi View according to the active view prior to selecting Mode View. The Left/Right cursor keys select different send pages in Strip View and the Up/Down cursor keys, the send slots in Multi View.

To change routing, if the new route is the same type as the current route, press and then rotate the encoder until the correct output is displayed, then operate the V-Sel switch to confirm the route. Routes will be displayed in the format 'Type' 'Bus No. Device' ie. 'Bus 1-2 U1'. The V-Pot is velocity sensitive to speed up scrolling through the destinations.

While scrolling through the routes Soundscape will also display the list of possible Sources and Destinations and the current position in a list on-screen, which provides full names and a much better overview.

Modifier Keys to globally or partially set Send Destinations (Channel or Multi View)

- ***Opt+*** V-Sel SWITCH – The current slot on all channels to the same destination
- ***Opt+Alt*** + V-Sel SWITCH – The current slot on all channels to incrementing destinations
- ***Shift+Opt+*** V-Sel SWITCH – The current slot on Selected channels to the same destination
- ***Shift+Opt+Alt*** V-Sel SWITCH – The current slot on Selected channels to incrementing source/destination/groups

## Sends Parameter Group Overview

| Channel Strip Functions | | | Parameter Group | | SENDS | | | | |
|---|---|---|---|---|---|---|---|---|---|
| **View** | **FLIP** | **Up/Down** | **Left/Right** | **Page** | **V-Pot** | **V-Sel** | **Fader** | **Cut** | **Notes** |
| Multi | Off | Send Slot | Parameter Pages | 1 | Send dB | Send Mute | Chan dB | Chan Mute | *Defaults to Send Slot 1* |
| | | | | 2 | Send Pan | - | Chan dB | Chan Mute | *Pan only on Stereo Sends* |
| | | | | 3 | - | Pre/Post | Chan dB | Chan Mute | |
| | On | - | - | - | Send Pan | Pre/Post | Send dB | Send Mute | *Pan only on Stereo Sends* |
| Channel Strip | Off | - | Send Pages | 1 | Send 1-8 dB | Send 1-8 Mute | Chan dB | Chan Mute | *Number of pages varies according the max number of sends in any mixer strip* |
| | | | | 2 + | Send 9-16 dB | Send 9-16 Mute | Chan dB | Chan Mute | |
| | On | - | Send Pages | 1 | Send 1-8 Pan | Send 1-8 Pre/Post | Send 1-8 dB | Send 1-8 Mute | *Pan only on Stereo Sends* |
| | | | | 2 + | Send 9-16 Pan | Send 9-16 Pre/Post | Send 9-16 dB | Send 9-16 Mute | |
| Mode | - | - | - | - | - | Route View | Chan dB | Chan Mute | *V-Sel 1 only. Displayed in each bay.* |
| Multi Route | - | Send Slot | | 1 | Destination Select | Flip Group / Confirm route | Chan dB | Chan Mute | *Destination Groups are: Bus, MADI, and Stream* |
| Channel Route | - | - | Send Pages | 2 + | Sends 1-8 Destination Select | Flip Group / Confirm route | Chan dB | Chan Mute | *Number of pages varies according the number of sends in any mixer strip* |
| | | | | | Sends 9-16 Destination Select | Flip Group / Confirm route | Chan dB | Chan Mute | |

## Pan

Selects V-Pots to control the mixer output panning

The V-Pots in Multi Channel View ere assigned to the first output pan parameter, which is Left/Right Pan/Balance for a mono or stereo channel, or the X position in a surround channel with an LCRS or 5.1 panner.

*NOTE: Any "xx to Mono" channels inside Soundscape will have no pan setting.*

The Left/Right cursor keys page the V-Pots through the additional pan parameters on channels with surround panners. V-Pots on channels with mono or stereo panners remain as pan controls.

| Parameter Page | 1 | 2 | 3 | 4 | 5 | 6 | 7 |
|---|---|---|---|---|---|---|---|
| Stereo | Pan | Pan | Pan | Pan | Pan | Pan | Pan |
| LCRS | X | Y | C% | F-DIV | R-DIV | F/R DIV | X |
| Surround | X | Y | C% | F-DIV | R-DIV | F/R DIV | LFE |

*NOTE: As with all parameter pages on Nucleus or Matrix, cursor keys wrap around to page 1, i,e, from Page 7...*

**FILP** assigns the current V-Pot pan parameter to the channel fader, and channel level on to the V-Pot.

In Channel Strip View (press PAN again) a single channel's pan parameters is assigned to the V-Pots in the selected channel's 8-fader bay (this is only really useful on surround channels).

| Parameter Page | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 |
|---|---|---|---|---|---|---|---|---|
| Stereo | Pan | - | - | - | - | - | - | - |
| LCRS | X | Y | C% | F-DIV | R-DIV | F/R DIV | - | - |
| Surround | X | Y | C% | F-DIV | R-DIV | F/R DIV | LFE | - |

**FILP** assigns the selected channels V-Pot pan parameters on to the channel faders, and the channel levels on to the V-Pot.

The temporary PAN Mode View (pressing and holding **PAN**) offers additional options for surround panning by assigning parameter pairs to the V-Pot and Channel Fader based on the Multi Channel View mode. Available pairings are X/Y, X/C%, and X/F-DIV.

**Pan Parameter Group Overview**

| Channel Strip Functions | | | Parameter Group | | PAN | | | | |
|---|---|---|---|---|---|---|---|---|---|
| **View** | **FLIP** | **Up/Down** | **Left/Right** | **Page** | **V-Pot** | **V-Sel** | **Fader** | **Cut** | **Notes** |
| Multi | Off | - | Parameter Pages | 1 | Pan (X) | - | Chan dB | Chan Mute | *See separate table for pages 2-7* |
| | On | - | Parameter Pages | 1 | Chan dB | - | Pan (X) | Chan Mute | |
| Channel Strip | Off | - | - | - | Pan Parameters | - | Chan dB | Chan Mute | *See separate table for V-Pot parameter assignment* |
| | On | - | - | - | Chan dB | - | Pan Parameters | Chan Mute | |
| Mode | - | - | - | - | Mode Select V-Pot 1–3 | - | Chan dB | Chan Mute | *Only on Surround Channels, Mono/Stereo channels remain in Mult Pan mode* |
| X/Y | - | - | - | - | X | - | Y | Chan Mute | *V-Pot 1* |
| X/C% | - | - | - | - | X | - | C% | Chan Mute | *V-Pot 2* |
| X/F-DIV | - | - | - | - | X | - | F-DIV | Chan Mute | *V-Pot 3* |

NAME/VALUE replaces the Pan Parameter label with the current pan parameter value regardless of view mode. When the **NAMES** are selected, the parameter value is displayed whilst a V-Pot is moved. The Parameter labels are restored after a short delay, which can be user defined in the SSL Console Control Configuration Dialogue.

## Plug-in

Selects V-Pots to control Plug-in parameters

In Multi Channel View after selecting the Plug-In Parameter Group Button displays the names of the first Plug-In inside mixer channels.

The up/down cursor keys cycle through additional plug-in's when available.

Channels without a plug-in at the currently selected position will display ' ------ ' above the V-Pot.

Cursor control of Plug-In No. selection wraps around as defined for the other parameter groups.

Current Plug-In No. is displayed in the consoles 2 Digit Mode Display.

The V-Sel key selects the plugin and assigns the plug-in parameters to the V-Pots inside all bays, that have been defined "Plug-in Spill Bays"(Plug-In checked) in the SSL Console Control Configuration Window.

The channel containing the plug-in becomes the selected channel as indicated by the channel strip SEL key.

Selecting Channel Strip View (by pressing the PLUG-IN Parameter Group button again) displays the names of the first eight plug-ins inside the selected channel in the control bay containing it.

After the last Plug-In only dashes ' ------ ' above the V-Pot indicate that there is no further Plug-In.

If there are mixer channels with more than eight plug-ins, the Left/Right cursor keys access additional Plug-In No. pages. As with Multi Channel View, the V-Sel key assigns the plug-in parameters to the Plug-in Spill Bays.

NAME/VALUE toggles the V-Pot label between Plug-In Number and Plug-in name.

**Plug-In Control**

Once a plug-in has been assigned to the V-Pots, the parameter names are displayed in the lower line of the V-Pot displays and the Channel Labels in the top row are replaced by the channel name and number of the strip with the selected plug-in, together with the plug-in name and the parameter page information (in the form current page/total number of pages).

*NOTE: The total number of pages will depend on the individual plug-ins and the number of Plug-In spill bays configured in the SSL Console Control Configuration.*

Turning a parameter V-Pot displays the current parameter value instead of parameter name, reverting to the name after the user definable delay.

Any dual state switched functions (eg. IN/OUT) in a plug-in will be controlled via the V-Sel switch and/or the V-Po.

NAME/VALUE toggles to display a full two-line display of the plug in with the Parameter name in the top row and parameter Value in the lower row.

The Left/Right cursor keys select the parameter pages, and the Up/Down cursor keys navigate to the next or previous plug-in of the selected channel automatically spilling the parameters for direct V-Pot control.

**FLIP** duplicates the plug-in parameters on to the faders allowing control from either the fader or V-Pot for non-switched functions.

The Channel **SEL** keys can be used to jump to the same plug-in No. in different channel, again with the parameters automatically spilled on to the V-Pots.

A subsequent operation of the **PLUG-IN** key, clears the spill bay and reverts to the last selected Plug-In view mode.

The temporary Plug-In Mode View menu offers two additional options, a specific configuration for controlling the SSL EQ via the faders and V-Pots in a single control bay when using the EQ key.

### Plug-in Parameter Group Overview

| Channel Strip Functions | | | Parameter Group | | PLUG-IN | | | | |
|---|---|---|---|---|---|---|---|---|---|
| **View** | **FLIP** | **Up/Down** | **Left/Right** | **Page** | **V-Pot** | **V-Sel** | **Fader** | **Cut** | **Notes** |
| Multi | - | Plug-in Slot | - | - | - | Plug-in Select | Chan dB | Chan Mute | |
| Channel Strip | - | - | Slot Pages | 1 | - | Slots 1-8 Plug-in Select | Chan dB | Chan Mute | *Slot pages dependent on Strip configuration* |
| Mode | - | - | - | - | - | 1- Chan EQ 2- Assign | Chan dB | Chan Mute | *See detailed description of Chan EQ mode* |
| Multi Assign | - | Plug-in Slot | - | - | Scroll Plug-in List | Plug-in Confirm | Chan dB | Chan Mute | |
| Channel Assign | - | - | Slot Pages | 1 | Scroll Plug-in List 1-8 | Plug-in Confirm Slots 1-8 | - | - | *Slot pages dependent on Strip configuration* |

### EQ and DYN keys

These two keys are active regardless of the current Parameter Group selection or View Mode, and will automatically open the first existing SSL EQ or SSL Channel Dynamics Plug-in on the currently selected channel strip, and assign the V-Pots in the Plug In Spill bay(s) ready for immediate control.

When the SSL EQ channel mode has been enabled from the temporary Plug-In Mode View menu, EQ controls are spilt in each bay for whichever channel was most recently selected in that bay. V-Pot labels are as shown in the second table and the plug-in spill bay(s) are not used.

| | 1 | 2 | 3 | 4 | 5 | 6 | 7 |
|---|---|---|---|---|---|---|---|
| **V-Sel** | In/Out | Input / Output | Bell / Shelf | Hz/Q Flip | Hz/Q Flip | Bell / Shelf | E- / G-Type |
| **V-Pot** | - | LF | Hz | Hz/Q | Hz/Q | Hz | HF |
| **Mute** | Bypass | - | - | - | - | - | - |
| **Fader** | In Gain | - | LF Gain | LMF Gain | HMF Gain | HF Gain | - |

Once enabled, using the EQ key will always spill the EQ controls in each bay for whichever channel was most recently selected in that bay. V-Pot labels are as shown in the second table and the plug-in spill bay(s) are not used.

### SSL- EQ Mode V-Pot Labels

| | **In / Out** | **Input / Output** | **Bell / Shelf** | **Hz / Q Flip** | **Hz / Q Flip** | **Bell / Shelf** | **E-Type / G-Type** |
|---|---|---|---|---|---|---|---|
| **NAME** | In Gain | HPF | LF | LMF | HMF | HF | LPF Hz |
| **VALUE** | dB | Hz | Hz | Hz/Q | kHz/Q | kHz | kHF |

When an EQ band gain fader is operated, the dB value is shown in the corresponding V-Pot label on the lower row regardless of the NAME/VALUE state. In NAME mode, V-Pot values are displayed in the lower row when a V-Pot is turned, reverting back to the label after a short delay.

## Channel/Track

Selects V-Pots to Control of Channel I/O Routing, Solo and Fader/Mute Group assignment

Selecting **CHANNEL** will initially focus the V-Pots on mixer strip Input Source assignment in Multi View.

**INPUT** will be displayed on the lower display above the V-Pot.

The left/right cursor keys are used select the alternative V-Pot Control options, Output Routing, and the Fader and Solo group assignment.

Changing Input and Output Routing operates as previously defined for Aux Sends. To change routing, press V-Sel and then rotate the encoder until the correct input or output port is displayed, then operate the V-Sel switch again to confirm the route. Routes will be displayed in the format 'Type' 'Bus No. Device' ie. 'Bus 1-2 U1'. The V-Pot is velocity sensitive to speed up scrolling through the sources or destinations. The V-Pot indicators will display.

While scrolling through the routes Soundscape will also display the list of possible Sources and Destinations and the current position in a list on-screen, which provides full names and a much better overview.

To change fader or solo group assignment, rotate the V-Pot until the required numbered group is displayed.

Fader and Solo Groups are only shown on screen, when Soundscape's Mixer is in WIDE View (X).

Switching to Channel View via a second operation of the CHANNEL switch displays the four channel setting function labels under V-Pots 1 – 4 in the bay containing the currently selected (SEL) channel. V-Pots 5-8 have no function and display '----' in the label area.

Holding **CHANNEL** selects the temporary Channel Mode View, which uses V-Sel 1-4 to directly select Multi View with the labelled function assigned to the V-Pots.

In either Channel Multi View or Channel Channel-Strip View, the modifier keys can be used to make global or selected channel routes and groups.

- ***Opt+*** V-Sel Switch – All channels to the same source/destination/group
- ***Opt+Alt+*** V-Sel Switch – All channels to incrementing source/destination/groups
- ***Shift+Opt+*** V-Sel Switch – Selected channels to the same source/destination/group
- ***Shift+Opt+ Alt*** V-Sel Switch – Selected channels to incrementing source/destination/groups

**CHANNEL/TRACK Parameter Group Overview**

| Channel Strip Functions | | | Parameter Group | | CHANNEL | | | | |
|---|---|---|---|---|---|---|---|---|---|
| **View** | **FLIP** | **Up/Down** | **Left/Right** | **Page** | **V-Pot** | **V-Sel** | **Fader** | **Cut** | **Notes** |
| Multi | - | - | Function Pages | 1 | Input Source | Flip Group / Confirm Route | Chan dB | Chan Mute | |
| | | | | 2 | Output Destination | Flip Group / Confirm Route | Chan dB | Chan Mute | |
| | | | | 3 | Fader Group | Confirm Route | Chan dB | Chan Mute | |
| | | | | 4 | Solo Group | Confirm Route | Chan dB | Chan Mute | |
| Channel Strip | - | - | - | - | 1- Input<br>2- Output<br>3- Fader Grp<br>4- Solo Grp | Flip Group / Confirm Route | Chan dB | Chan Mute | *Routing and Group control for selected channel(s)* |
| Mode | - | - | - | - | - | 1- Input<br>2- Output<br>3- Fader<br>4- Solo | Chan dB | Chan Mute | *Fast access to Multi View options* |

## Modifier Keys

**SHIFT, CONTROL, OPTION, ALT**

These keys modify the actions of other functions keys as described in this overview and detailed in the associated function tables above.

SHIFT, CONTROL and ***Alt*** parallel their equivalents on a standard QWERTY PC keyboard, OPTION is part of the MCU protocol and a valid key found on MAC keyboards. The modifier keys may be used on their own or in conjunction with other modifier keys.

## Mixer Channel Navigation

Two sets of keys, **CHANNEL** < / > and **BANK** < / >, control the relationship between the physical control surface channel strips and the columns in Soundscape's Mixer.

Initially the surface strips are assigned to ascending channels starting from Column 1 in Soundscape's Mixer.

The **CHANNEL** keys shifts the surface focus by single channel increments in the indicated direction, and the **BANK** keys shifts the surface focus by a fixed channel amount (in multiples of eight) that can be user defined in the SSL Console Control Configuration Dialogue.

| | |
|---|---|
| **CHANNEL** < ***+Shift*** | Left justifies the mixer - Soundscape Mixer Column 1 is assigned to the first strip on the surface with ascending channels assigned to the remaining strips. |
| **CHANNEL** > ***+Shift*** | Right justifies the mixer - The last column in the Soundscape mixer is assigned to the rightmost strip on the surface with descending channels assigned to the remaining strips, from right to left. |
| **BANK** < ***+Shift*** | Same as **CHANNEL** < ***+Shift*** |
| **BANK** > ***+Shift*** | Assigns the banks containing the last channels in the Soundscape mixer to the controller, This is dependant on the size of the mixer, the channel bank size and the number of faders in the control surface. |

*NOTES:*
*The active bays on the Console are shown in Soundscape's Mixer Column Number bar with a dark red background. The lowest indexed (or left most) Mixer column labelled "MASTER", "Master" or "master" in Soundscape's mixer is not included in CHANNEL or BANK operations and is always controlled from the dedicated Controller Master fader as well as the normal console channel, if the Console supports this and the MASTER option in the SSL Console Control Setting is activated.*

## Channel Strip Controls

| | |
|---|---|
| **SEL** | Used to nominate the strip for various channel parameter operations detailed in the V-Pot Operation description. Normally only a single strip can be selected and the **SEL** keys intercancel. However Multiple strips can be selected in conjunction with the Modifier Keys. The current selected strip can be deselected via a subsequent operation of the **SEL** key. Selected Mixer Strip is indicated on screen (mechanism to be determined) and via illuminating the **SEL** key tally. |
| **SEL *+Shift*** | Selects all channels between the currently selected strip and the strip accessed via the **SEL *+Shift*** operation. |
| **SEL *+Alt*** | Allows multiple non contiguous selection |
| **SEL *+OPT*** | If a single strip selected, selects all strips. If multiple strips selected, cancels current strip selection. |
| **REC**(**SEL**) | The editor audio track arming mode is enabled via the master REC ARM or REC key on the Control Surface. Once enabled the **SEL** keys output the MCU key codes corresponding to the REC/RDY function. The Track arm command enables the Record function on the Editor channel assigned to the Soundscape mixer Strip. |
| **REC *+Opt*** | Record enable all Tracks, or Global Cancel Record if tracks already enabled. |
| **REC *+Opt+Shift*** | Record enable selected Tracks which have been previously selected via **SEL *+Shift*** or **SEL *+Alt***. |
| **REC *+Alt*** | Record Enable Mixer Strip Automation Track |
| **REC *+Opt*** | Record Enable All Mixer Strip Automation Tracks, or Global Cancel Record if tracks already enabled. |
| **REC** *+**Alt+Opt+Shift*** | Record enable Mixer Automation on Tracks which have been previously selected via **SEL *+Shift*** or **SEL *+Alt***. |
| **SOLO** | Activates the Mixer Strip solo function. |
| **SOLO *+Opt*** | Global solo on all tracks, or cancels active solo's on all tracks. |
| **SOLO *+Ctrl*** | Activates Solo on all Channels assigned to the same Solo Group |
| **SOLO *+Opt+Shift*** | Activates Solo on all Channels which have been previously selected via **SEL *+Shift*** or **SEL *+Alt***. |
| **MUTE** | Activates the Mixer Strip Output mute function. |
| **MUTE *+Alt*** | Activates the Mixer Strip Input mute function. |
| **MUTE *+Opt***(/***+Alt***) | Global Output mute on all tracks, or cancels active mutes on all tracks. |
| **MUTE *+Ctrl*** (/***+Alt***) | mutes All Channels assigned to the same Fader Group. |
| **MUTE *+Opt+Shift*** | mutes all Channels which have been previously selected via **SEL *+Shift*** or **SEL *+Alt***. |
| Fader ***+Ctrl*** | All Faders assigned to the same Fader Group follow the touched fader. The link is a relative dB link and not just positional. |
| V-POT/V-SEL | See V-Pot Operation for detailed description. |

## Displays

**Counter:** 10 digit (3+2+2+3) numeric display indicating current timeline position. Two readout options, Timecode, -HH.MM.SS.FF-, or Bar(4)/Beats(2)/Sub Beats(3) -BB.BB.BB.SSS. Readout format is toggled with the SMPTE/BEATS key and two dedicated front panel tallies indicate SMPTE or Bar/Beats.

**SMPTE/BEATS** Toggles Counter readout format.

**DAW Solo** or **Solo** Front panel Tally active if any mixer channel is in SOLO

## Function Keys

16 Functions Keys in conjunction with the modifier keys directly assign editing tools to the left and right mouse keys. To connect the Function keys to Soundscape Tools create a new Tool Page inside Soundscape with the order of the first 16 Editing Tools reflecting the F1-16 keys. This Tool Page (ie. Page 5) can be selected in the SSL Console Control Configuration Window with the MAP parameter (select 5 for Tool Page 5).

F-KEY(N) Assigns the loaded tool to the Left Mouse click

F-KEY(N) ***+Ctrl*** Assigns the loaded tool to the Right Mouse click

F-KEY(N) ***+Alt*** Assigns the loaded tool to the ***Alt*** Left Mouse click

F-KEY(N) ***+Ctrl+Alt*** Assigns the loaded tool to the ***Alt*** Right Mouse click

## Automation Control

**AUTO ENABLE** Mixer Automation enable equivalent to the onscreen box or G keyboard shortcut

**TOUCH** Enables 'Touch record' in the Automation Setup menu. Default mode is Normal Record

**TOUCH TO STOP** Inter-cancels with TOUCH and enables 'Touch record till stop' in the Automation Setup menu. Default mode is Normal Record

**CYCLE END** Toggles the 'Apply last recorded value from Part begin' option in the Automation Setup menu

**TAKE SNAPSHOT** Equivalent to the on screen one shot 'Snapshot Automation Record' Function or F keyboard shortcut

## General File/Edit Commands

**SAVE** Save Arrangement

**SAVE*+Shift*** Save Mix

**UNDO** Undo (as found in the Edit Menu)

**UNDO*+Shift*** Redo (as found in the Edit Menu)

*The button tallies indicate if files are in an unsaved state or Undo/Redo's are available.*

## Timeline/Transport Control

| | |
|---|---|
| **REW** | Rewind |
| **FF** | Fast Forward |
| **STOP** | Stop |
| **PLAY** | Play |
| **REC** | Master Record |
| **STOP** + **STOP** | Go to beginning |
| **PLAY** + **PLAY** | Play from Previous Play Location (Repeat Play) |
| **REW** ***+Shift*** | Go to Previous Part |
| **FF** ***+Shift*** | Go to Next Part |
| **CYCLE** | Loop mode On/Off |
| **NUDGE–** | Nudge back 1 Frame or Beat according to SMPTE/BAR BEAT Mode |
| **NUDGE+** | Nudge forward 1 Frame or Beat according to SMPTE/BAR BEAT Mode |
| **NUDGE–** ***+Shift*** | Nudge back 1 Sec or Bar according to SMPTE/BAR BEAT Mode |
| **NUDGE+** ***+Shift*** | Nudge forward 1 Sec or Bar according to SMPTE/BAR BEAT Mode |
| **SET LEFT** | Set Left Locator to current timeline position |
| **SET RIGHT** | Set Right Locator to current timeline position |
| **SET LEFT** ***+Shift*** | Locate to Left Locator position |
| **SET RIGHT** ***+Shift*** | Locate to Right Locator position. |
| **SET LEFT** ***+Alt*** | Set Punch In to current timeline position |
| **SET RIGHT** ***+Shift*** | Set Punch Out to current timeline position |
| **SNAP** | Toggles the Timeline Snap Mode |

## Markers

**SET MARK** Creates a Mark at the current timeline position. Equivalent to the PC Keyboard 'Insert' Key function

Three keys, **NEXT**, **LAST** and **PREVIOUS**, are used to navigate the marker list. Correct operation of these functions is conditional on the Marks list being created in ascending positional order. Initially **NEXT** outputs Marker ID 1 in the list. Subsequent operations of the **NEXT** key select successive incrementing Marker ID's.

**PREVIOUS** outputs the Marker ID for the Marker one below last **NEXT** Marker ID used. Subsequent **PREVIOUS** commands continue to decrement the Marker ID sent. **LAST CUE** repeats the last Marker ID output by either the **NEXT** or **PREVIOUS** key depending on which one was last used.

## Cursor Keys

| | |
|---|---|
| **LEFT**/**RIGHT**/**UP**/**DOWN** | Primarily used for V-Pot parameter navigation as previously described in the V-Pot Operation overview. In conjunction with the **ZOOM** switch can also control the Arrange View Zoom functions |
| **ZOOM** | Switches the cursor keys to control the Arrange Window track view. When active, **UP** and **DOWN** control Track height and in turn, the number of visible tracks. **LEFT** and **RIGHT** control Track width (Horizontal Zoom) |

## Jog Wheel

The Jog Wheel is used to scroll the current timeline position, ie. the current locator cursor. The Speed/Resolution is zoom level dependant, ie. faster when zoomed out.

| | |
|---|---|
| **SCRUB** | Selects the Audio Scrub Mode controlled by the Jog Wheel |

## Foot Switch Control

Two momentary switch inputs to control transport functions.

| | |
|---|---|
| INPUT 1 | Toggles Transport play/stop; ie. if Transport is stopped, first operation initiates play, and a subsequent operation stop |
| INPUT 2 | Toggles Transport record, allowing remote punch in/out control |

# MCU (Mackie Control Universal) command Mapping to Soundscape

| Page 1 | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|
| **MACKIE Control Logic Mapping** | | **Universal Mapping** | **SOUNDSCAPE** | **Modifiers** | | | | | | | |
| Sect'n | Sub Sect'n | Function | Equivalent | SHIFT | ALT | OPT/ALL | CTRL | OPT + SHIFT | ALT + CTRL | OPT + ALT | SHIFT+ OPT + ALT |
| Master Strip | Assignment | TRACK | CHANNEL | | | | | | | | |
| | | PAN/ SURROUND | PAN | | | | | | | | |
| | | EQ | EQ | | | | | | | | |
| | | SEND | SEND | | | | | | | | |
| | | PLUG IN | PLUG-IN | | | | | | | | |
| | | INSTRUMENT | DYNAMICS | | | | | | | | |
| | Fader Banks | BANK < | BANK < | GOTO BANK 1 | | | | | | | |
| | | BANK > | BANK > | GOTO LAST BANK | | | | | | | |
| | | CHANNEL < | CHANNEL < | LEFT JUSTIFY | | | | | | | |
| | | CHANNEL > | CHANNEL > | RIGHT JUSTIFY | | | | | | | |
| | | FLIP | FLIP | | | | | | | | |
| | | GLOBAL VIEW | SSL TOOL PAGE | | | | | | | | |
| Channel Strip | | REC/RDY | REC/RDY | | AUTO TRK ARM | GLOBAL REC ARM | | ARM SELECTED | | GLOBAL AUTO REC | SELECTED AUTO REC |
| | | V-SEL | V-SEL | | | | | | | | |
| | | V-POT | V-POT | | | ALL V-POT | | SELECTED V-POT | | | |
| | | SOLO | SOLO | | | ALL SOLO | SOLO GROUP | SOLO SELECTED | | | |
| | | MUTE | MUTE (OUTPUT) | INPUT MUTE | | ALL MUTE | MUTE GROUP | MUTE SELECTED | | | |
| | | SELECT | SELECT | | | ALL SELECT | MULTI SELCT | | | | |
| | | FADER (TOUCH) | FADER | | | | MOVE GROUP | | | | |
| | | SOLO IND | SOLO IND | | | | | | | | |
| Misc | Foot Switch 1 | STOP/PLAY | STOP/PLAY | | | | | | | | |
| | Foot Switch 2 | RECORD | REC IN/OUT | | | | | | | | |

| Page 2 | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|
| **MACKIE Control Logic Mapping** | | **Universal Mapping** | **SOUNDSCAPE** | **Modifiers** | | | | | | | |
| Sect'n | Sub Sect'n | Function | Equivalent | SHIFT | ALT | OPT/ALL | CTRL | OPT + SHIFT | ALT + Ctrl | OPT + ALT | SHIFT+ OPT + ALT |
| Master Section | Display | NAME/ VALUE | NAME/VALUE | | | | | | | | |
| | | SMPTE/BEATS | SMPTE/BEATS | | | | | | | | |
| | Function Keys | F1 | Tool 1 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | | F2 | Tool 2 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | | F3 | Tool 3 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | | F4 | Tool 4 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | | F5 | Tool 5 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | | F6 | Tool 6 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | | F7 | Tool 7 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | | F8 | Tool 8 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | Global Views | MIDI TRACKS | Tool 9 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | | INPUTS | Tool 10 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | | AUDIO TRACKS | Tool 11 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | | AUDIO INSTRUMENTS | Tool 12 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | | AUX | Tool 13 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | | BUSSES | Tool 14 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | | OUTPUTS | Tool 15 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | | USER | Tool 16 | | Alt Left Switch | | Right Mouse Sw | | Alt Right Sw | | |
| | Modifiers | SHIFT | SHIFT | | | | | | | | |
| | | OPTION | OPTION | | | | | | | | |
| | | CONTROL | CONTROL | | | | | | | | |
| | | ALT | ALT | | | | | | | | |
| | Automation | READ/OFF | AUTO ENABLE | | | | | | | | |
| | | WRITE | TOUCH | | | | | | | | |
| | | TOUCH | TOUCH TO STOP | | | | | | | | |
| | | LATCH | CYCLE END | | | | | | | | |
| | | TRIM | TAKE SNAPSHOT | | | | | | | | |

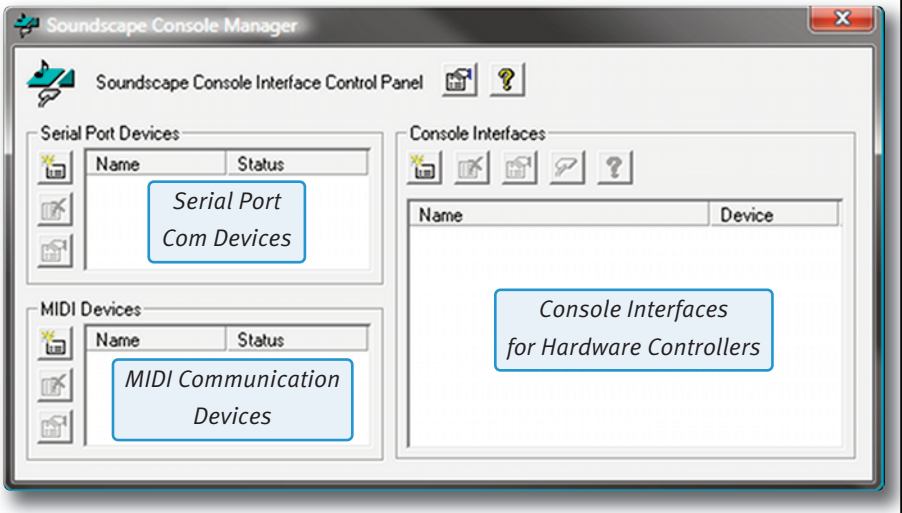
| Page 3 | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|
| **MACKIE Control Logic Mapping** | | **Universal Mapping** | **SOUNDSCAPE** | **Modifiers** | | | | | | | |
| Sect'n | Sub Sect'n | Function | Equivalent | SHIFT | ⍰ALT | OPT/ALL | CTRL | OPT + SHIFT | ALT + CTRL | OPT + ALT | SHIFT+ OPT + ALT |
| Master Section (cont) | Functions | SAVE | SAVE | SAVE MIX | | | | | | | |
| | | UNDO | UNDO | REDO | | | | | | | |
| | | GROUP | SNAP | | | | | | | | |
| | | CANCEL | LAST MARK | | | | | | | | |
| | | ENTER | SET MARK | | | | | | | | |
| | Locate Functions | MARKER | SET LEFT | GOTO L | SET REC IN | | | | | | |
| | | NUDGE | SET RIGHT | GOTO R | SET REC OUT | | | | | | |
| | | CYCLE | CYCLE | | | | | | | | |
| | | DROP | NUDGE - (FRM/BEAT) | SEC/BAR | | | | | | | |
| | | REPLACE | NUDGE + (FRM/BEAT) | SEC/BAR | | | | | | | |
| | | CLICK | PREVIOUS MARK | | | | | | | | |
| | | SOLO | NEXT MARK | | | | | | | | |
| | Transport Keys | REWIND | REW | GOTO PREVIOUS PART | | | | | | | |
| | | FF | FF | GOTO NEXT PART | | | | | | | |
| | | STOP | STOP | | | | | | | | |
| | | PLAY | PLAY | | | | | | | | |
| | | REC | REC | | | | | | | | |
| | Cursor Buttons | < | < | | | | | | | | |
| | | > | > | | | | | | | | |
| | | UP | UP | | | | | | | | |
| | | DOWN | DOWN | | | | | | | | |
| | | ZOOM | ZOOM | | | | | | | | |
| | Jog Wheel | SCRUB | SCRUB | | | | | | | | |

# Legacy Console Manager

## Console Manager Properties Window

If you have installed Console Manager (see Chapter 2) Click on Properties or double click the task bar icon to open the main Console Manager Properties Window

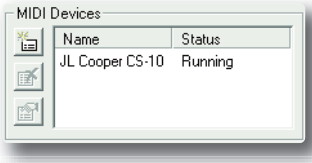
Console Manager allows external devices to control the SSL Mixer software by receiving data from these external devices and transmitting it to the Mixer Software, using several possible Control Protocols (HUI, MCU, JL Cooper etc.) that can be set up in the Console Interface section on the right hand side of the window.




The two sections on the left are used to select the Hardware Ports (Serial Com Ports, Midi Interface, USB MIDI) that belong to the Hardware Controller or where the Controller is connected to (MIDI, RS422 etc.).

## Setting Up Midi Controller Devices

The MIDI Devices section has three buttons: **Add**, **Remove**, and **Properties**. The latter two appear greyed-out if no Midi device is selected or present.



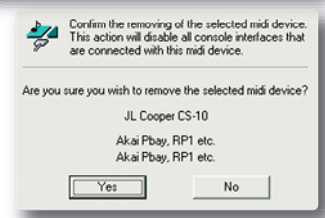



Clicking the **Add button** opens the dialogue to create a new Device. Select any available MIDI Input and Midi Output the device is connected to, and give the device a memorable name you can remember.






Clicking the **Remove button** when a device is selected opens a dialogue box where the removal of that device must be confirmed. Click Yes to remove the device, click No to abort the operation.

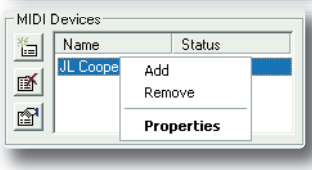





Clicking the **Properties button** when a device is selected opens a dialogue box which shows the properties of that device, as it was defined upon creation.

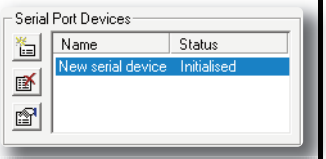



Right Clicking the entry of a MIDI device opens a menu with the same functions as the three buttons described above.



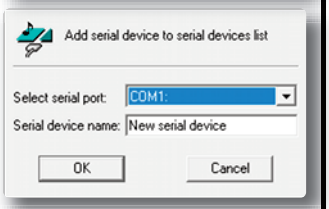
## Setting Up Serial Com Controller Devices

The Serial Devices section has three buttons, **Add**, **Remove**, and **Properties**. The latter two appear greyed-out if no Serial Com device is selected or present.






Clicking the **Add button** opens the dialogue to create a new Device. Select any available Serial Comport Device (RS232 or RS 422) the external device is connected to, and give the device a memorable name you can remember.






Clicking the **Remove button** when a device is selected opens a dialogue box where the removal of that device must be confirmed. Click **Yes** to remove the device, click No **to** abort the operation.

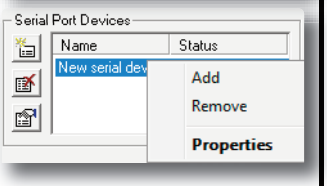





Clicking the **Properties button** when a device is selected opens a dialogue box which shows the properties of that device, as it was defined upon creation.

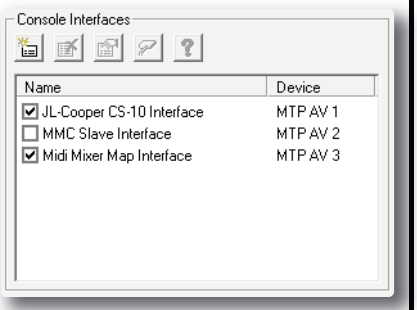



Right Clicking the entry of a Serial Port device opens a menu with the same functions as the three buttons described above.




## Setting up Console Interfaces

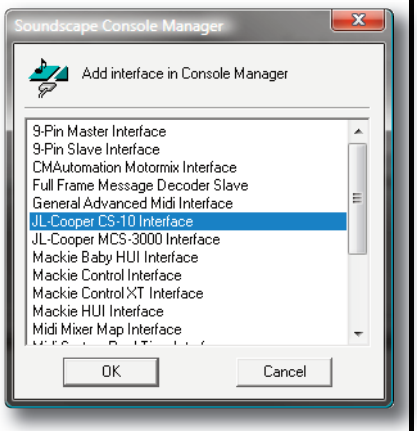
The Console Interfaces section has five buttons, **Add**, **Remove**, and **Properties**, **Change Device** and **Information.** The latter four appear greyed-out if no Console Interface device is selected or present.



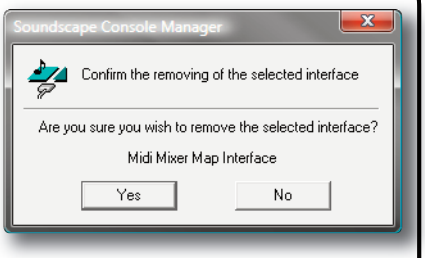



Clicking the **Add button** opens the dialogue to select the kind of device you have connected. The entries in the Interface list partly refer to industry standard communication protocols and partly to dedicated Hardware devices. Some of the dedicated Hardware devices, however, defined industry standard communication Protocols. Selecting for example a MACKIE HUI interface also works with other Hardware, that supports the MACKIE HUI (HUI) protocol.

Select the appropriate Protocol/Device Interface and press **OK**.

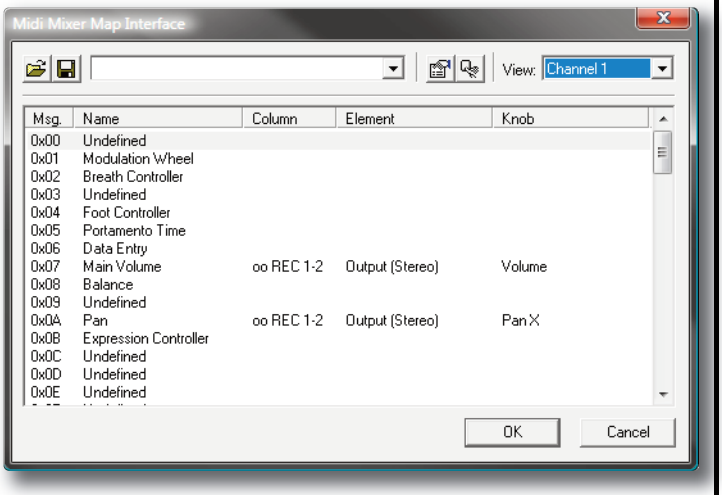


Clicking the **Remove button** when an Interface is selected opens a dialogue box where the removal of that device must be confirmed. Click Yes to remove the device, click No to abort the operation.



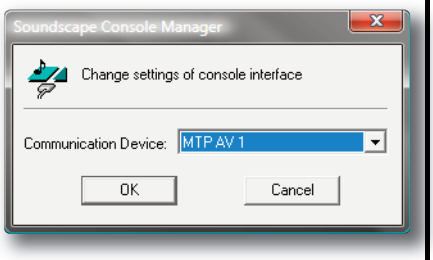

When certain Console Interfaces such as the MIDI Mixer Map Interface or Soundscape 9-Pin Slave Interface are selected, clicking the **Properties button** opens a dialogue box which shows information about the interface and allows editing of the setup.

This window also may include access to further windows, like the MIDI 'Learn' mode etc.

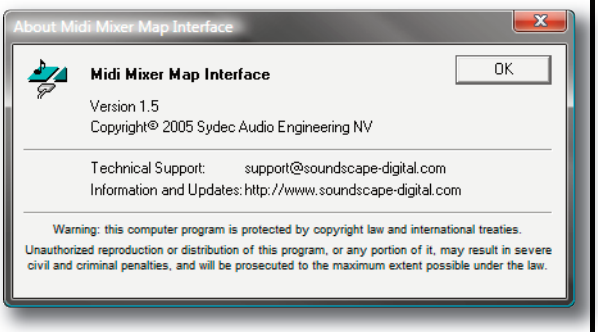



Clicking the **Change Device button** when a Console Interface is selected opens a dialogue box where any of the installed Serial Ports and MIDI devices can be selected and assigned to the selected Console Interface:

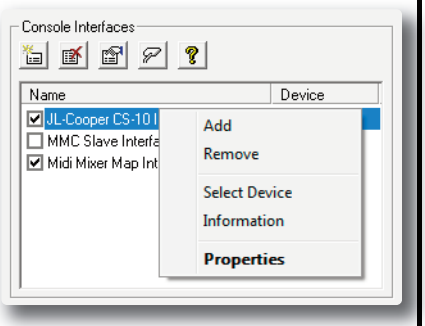
*NOTE: While any controller device can be selected from the list, the selected Console Interface will only function correctly with a compatible device.*



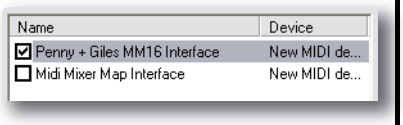

Clicking the **Information button** opens an information window about the selected Console Interface.




Right-clicking on a Console Interface opens a menu with five entries that match the functions of the five buttons just discussed.



## Activating a Console Interface




When the Console Interface is created and assigned to an appropriate Communication Device (Serial or MIDI) and the Hardware Controller is connected to the selected communication device, you can activate the Console Interface by checking the tick box belonging to the Interface entry. (Penny + Giles MM16 Interface in the screen shot on the right).

*NOTE: While several Console Interfaces can be used simultaneously to control the same Mixer, certain configurations create conflicts and different Hardware Controllers start to 'fight' for the right Fader Position.*

## Saving a Console Manager configuration

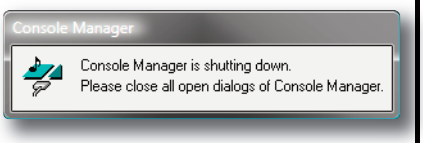
Activating the option Connect to Console Manager in the menu: **Settings** / **Preferences** / **Mixer** initiates the basic communication between Soundscape and the Console Manager Module.

Performing a **Save Settings** (from the **Settings** menu) while Console Manager is running and connected makes the connection permanent, and Console Manager will be launched automatically the next time the SSL Mixer is started.

Console Manager's settings are saved automatically whenever you change any setting and will automatically run the same configuration on next start up. This includes the selection of Communication devices, created Console Interfaces, devices selection and Activation Status of the Console Interfaces.

*NOTE: When Console Manager starts up (together with launching the SSL Mixer) it scans the computer for available Communication Devices. If you for example have a USB Hardware Controller, that is still switched off, Console Manager can not initialise the MIDI Port(s). In order to remedy the situation you should re-launch the Mixer software after you have switched on the Controller.*

## Closing Console Manager



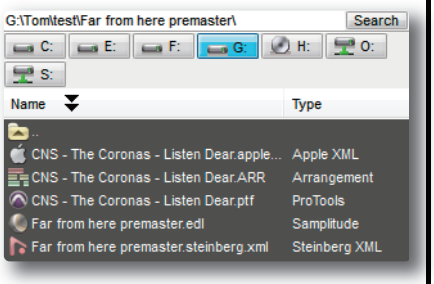

Console Manager will automatically close at the same time as the Soundscape software, provided that no Console Manager windows and/or dialogue box is still opened, in which case the following dialogue box will appear.

The same dialogue box will appear, when you disable **Connect to Console Manager** in the Mixer Software, and Console Manager windows are still open.

# File Manager

Soundscape's File Manager is the central place to access any kind of relevant file available to your PC. In a similar way to Windows Explorer, it handles the Arrangements, Mixes, Audio Takes and Automation Takes used by Soundscape.

The File Manager can either be docked with the Arrange window, or be displayed in its own window, as defined by the **View** menu's **Dock Arrange Window & File Manager** option (also toggled by the ***K*** key). When docked, the scroll bar between the Arrange and File Manager areas can be dragged to change the proportion of the window given to each. When un-docked, its window can be re-sized in the normal way, and can be opened or closed via the **View** menu's **File Manager** option (or using the ***D*** key).




Soundscape doesn't have any concept of a Bin, Media Pool or the necessity to Import directly readable audio files into a collection. The File Manager uses simple drag-and-drop to add audio or automation to the Arrangement, or to append Mixer files to the Mixer. There is also no concept of Regions or anything similarly confusing in the File Manager.

There are special icons for each kind of folder and file and their active/inactive status that show, if files are used inside the Arrangement.

File Manager also has very useful options for copying/moving files, collecting all project files and creating project back-ups easily and safely.

To open an Arrangement or Mixer file, double-click on it. The Arrangement and Mix icon will be highlighted in blue and **Arrangement (Act)** and **Mix (Act)** (for Active) will be shown in the **Type** column.

Audio Files are simply dragged into the Arrange Window; If necessary, a file conversion or sample rate conversion can be done in the background.

Icons of Takes that are referenced by the Parts in the Arrangement are highlighted in green (24-bit audio) or in blue (16-bit audio and automation Takes).

Audio files and even complete Arrangements can be auditioned directly in File Manager.

---

*Note that WAV and BWF files can be dragged into an Arrangement directly from Windows Explorer.*

---

## Overview and Navigation

### Current Path

The current path is displayed in the top of the File Manager. If the path is coloured, the opened folder is marked as the current Record/Process Folder (red letters) or is part of a Domain/Root folder (blue letters).




### File List

The file list shows all relevant information for files and folders and allows all sorts of administration tasks like renaming, moving, copying, pasting, deleting, etc.

Clicking on a file attribute label allows defining the sorting priority. Clicking for a second time on an item will toggle between ascending and descending order. Each column can be resized by dragging its left or right borders. Columns can also be re-ordered by dragging any of them to a new position.

## File List Sorting

Left-clicking in the header area of a file list column will sort the list by that column; left-clicking again will reverse the file order. The list is pre-sorted into two groups:

1 folders appear at the top of the list,
2 files appear at the bottom of the list.

Right-clicking in the header area will sort the list by that column, while pre-sorting the list into three groups:

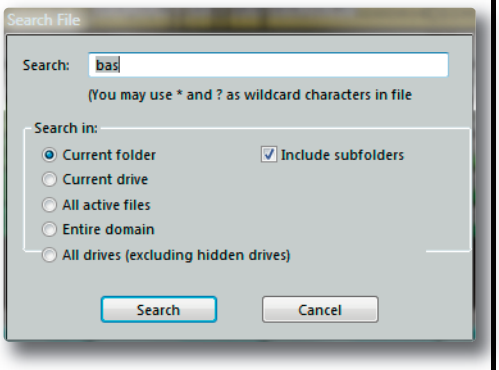
1 folders appear at the top of the list,
2 followed by Arrangements, mixers, and any other file that is not a Take or audio file,
3 with Takes and audio files placed at the bottom of the list.

Another right-click will reverse the sorting, retaining the three-way pre-sorting.

Arrows in the header area indicate the current sort column, with the arrow direction indicating the sort direction. Single arrows indicate two-way pre-sorting, and double arrows indicate three-way pre-sorting.

## File Searching

To search for a specific file, click the **Search** button at the right end of the Current Path display to open the **Search File** dialogue. Simply type the word or parts of the word you want to search for, define the scope of the search, and hit **Search**. The search results are displayed in the File Manager, as indicated by the text in the Current Path display. To audition Audio/Automation files, drag them to the Arrange Window, or double click Arr and Mix files to open them.




To return to the standard folder view inside the last displayed folder, click once on the active Drive button in the Drive Display.

*Note that the search function can also be accessed by selecting **Search File...** which is found in the **File** menu and in the File Manager right-click menu.*

## Drive Display

In the drive display, all logical drives attached to your system are shown and the opened drive button is illuminated. The display remembers the path of the last opened folder for each drive. Different icons are shown for internal HDDs, external HDDs (USB/eSata/Firewire), removeable Media(CD/DVD/Tape) and mapped Network Drives.




*Note that the File Manager only Lists Drives that have a Drive Letter assigned in Windows Disk Manager.*

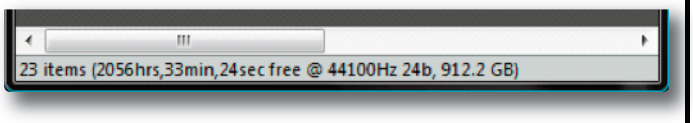
Soundscape only scans for drives on startup and does not automatically update the Drive Bar when an external Drive is plugged or unplugged. To update the display, Soundscape needs to be restarted. To reload the content of a drive (in case it has changed in the background) simply click on the active drive's button.

## Hiding Drives

To hide Windows drives from the drive display (eg. DVD Burner, System Drive), go to the **Settings** / **Preferences** / **File Manager** / **Hide Logical Drive** menu and select a drive letter (from A: to Z:) in the final submenu. Hiding and showing drives is immediately reflected in the display. Active Takes from hidden drives remain playable by referencing Parts in the Arrangement.

## Status Bar

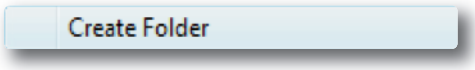
The status bar at the base of the window shows the number of files present in the folder or number of selected files, with the free disk space shown in brackets (in GB and in mono track time at the current sample rate) on that drive.



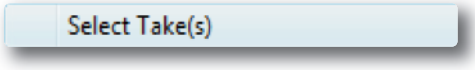
## File and Folder Operations

The File Manager window uses context sensitive interaction, using the left and right mouse buttons. A left double click on any blank area of the file list, or on the icon (or in case the folder is part of a Domain/Root), takes you up one level in the folder tree. Left double click on a folder to open it. Right clicking on a file, folder or a blank area of the file list opens different context sensitive menus:

### Right-Clicking on an Empty Space




The **Create Folder** command works in a very similar way to Windows Explorer. When a folder is created it is ready to be named.



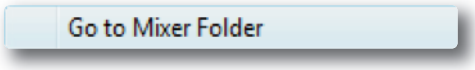

If Parts are selected in the Arrange window, the **Select Take(s)** option allows selecting the Take(s) referenced by the selected Part(s). This option is only available if all the Takes referenced by the selected Parts are inside the same Folder. If the option is not available it appears dimmed in the pop-up menu.



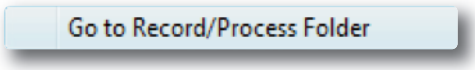

Selecting **Search File...** opens a pop-up in which you can define a search for a file, as described on the previous page.



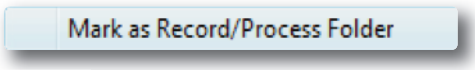

The **Go to Arrangement Folder** command opens the folder where the currently active Arrangement is located.




The **Go to Mixer Folder** command opens the folder where the currently active Mixer is located.



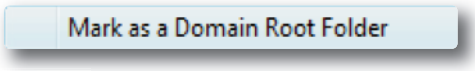

The **Go to Record/Process Folder** command the selected Record/Process Folder for the active Arrangement.





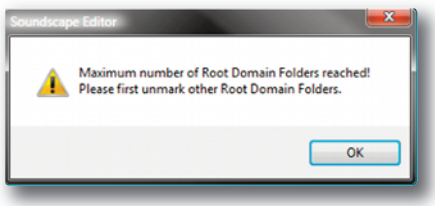

The **Mark as Record/Process Folder** command assigns the open Folder as Record/Process Location, where the Active Arrangement will store recorded and processed takes. This folder can be reassigned to any folder by right clicking on it and selecting the **Mark as Record/Processing Folder** option.

This folder setting is saved in the Arrangement file so every time it's opened, the location to store audio, automation, mix and arrangement files is remembered.






Selecting the **Mark as Domain Root Folder** command marks the selected or currently open folder as 'Domain Root Folder'. The Domain Root folders allow Soundscape to minimize the time to search for take files when an arrangement is opened. Audio Takes are then only searched within these specified folder paths (including subfolders).

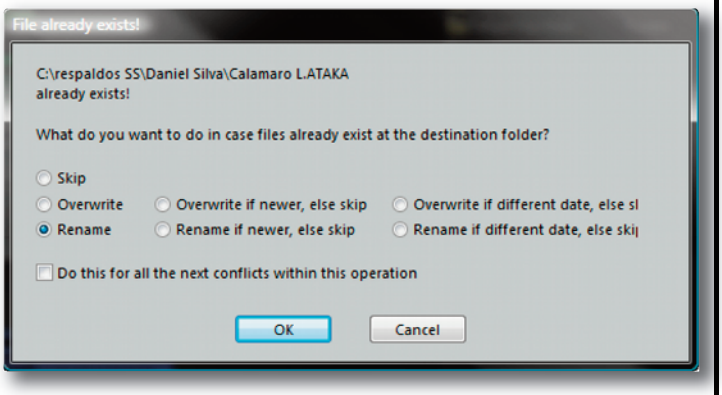



Up to 8 'Domain Root' folders located on any logical drive can be specified. Once you attempt to assign the 9th Domain Root, an error dialogue will appear:

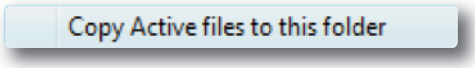
*Please note that marking a subfolder of a Domain Root Folder, removes the status from the parent folder.*



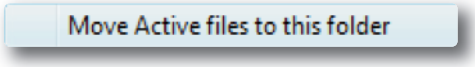

Pasting is a straight forward process. However, if required, Soundscape may need your input for pasting or copying files that already exist inside the destination directory:




- **Skip** will not copy/paste the file;
- **Overwrite** will overwrite existing files with the same name;
- **Rename**: pasted files will be renamed with a numerical extension (2, 3...) appended to their name;
- **If Newer** option overwrites or renames existing files only if the pasted file is newer (older files are skipped);
- **If different Date** option overwrites or renames existing files only if the pasted file has a different date (equal dates are skipped);
- **Do this for all conflicts** tick box applies your selected resolution to any other filename conflict.




The **Copy Active files to this folder** command will copy the files used be the active arrangement to the currently open folder. This is specially useful if the audio and automation takes used by the arrangement come from different folders (ie. some of them are sound effects from an SFX storage drive) and you want to back-up or share the arrangement. This command will also copy the currently active mix and arr files to the folder.

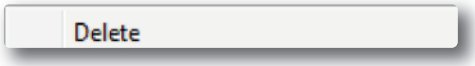



Selecting the **Move Active files to this folder** command will move the files used be the active arrangement to the currently opened folder. This an easy way to gather all the audio and automation takes and the currently active mix and arr files into a single folder.

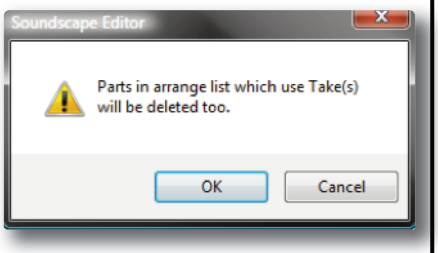
## Right clicking on a folder

Right clicking on a folder opens a different context sensitive menu.

**Rename**, **Cut** and **Copy** work in the same way as in Windows Explorer. The **Mark as Domain Root Folder** function is also present in this menu.

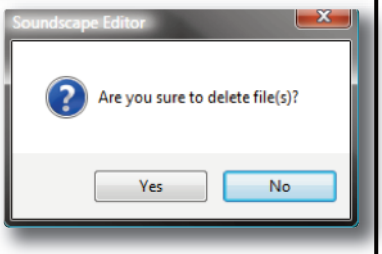



Right click and select **Delete** on any Folder in the File Manager window to delete it. If there are Parts in the Arrange window that reference Take(s) inside this folder, a dialogue box will be displayed.



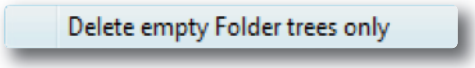

Click on **Yes** only if you want to continue with deleting the Take(s).

If any Part(s) in the current Arrangement reference any Take(s), a further dialogue box appears that warns you that this (these) Part(s) will be deleted.




It is also possible to delete Folders or any other files from the File Manager window by highlighting the relevant file and then using the ***Delete*** key.

*NOTE: Be very careful when deleting Files. You can undo the delete and there are up to 99 levels of undo. However, if you perform several operations after deleting Files, it is not certain that you will be able to recover them. It depends if they have been physically overwritten on the disk. Files deleted in Soundscape File Manager are not put into Windows Recycle Bin.*



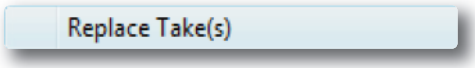

By selecting **Delete empty Folder trees only,** all empty folders and subfolders within the currently selected folder are deleted.

## Right clicking on a Take File

Right clicking on an audio Take (ATAK, WAV or BWF) file opens another menu.

**Rename**, **Cut** and **Copy** behave in the same way as you would expect in Windows Explorer.

**Delete, Delete empty Folders trees only** have been explained above.



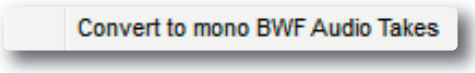

With the **Replace Take(s)** command, you can replace the Take references of one or more selected Parts (in the Arrange Window) with the selected Take (in the File Manager).

This option is only available if the selected Take is at least as long as the longest selected Part and is of the same type as all the selected Parts (Audio or Automation).

Please note that the Part's **Start in Take** offset parameter (as shown in the **Part Info** window) is maintained, and the selected Take must also be long enough to allow for that offset.

If **Replace Take(s)** is dimmed/greyed out, the original Part is longer than selected new Take.

The selected Part(s) will also be renamed using the new Take's name if the **Replace Part name(s) as well when replacing Take(s)** option is selected under **Settings** / **Preferences** / **Arrangement** menu.

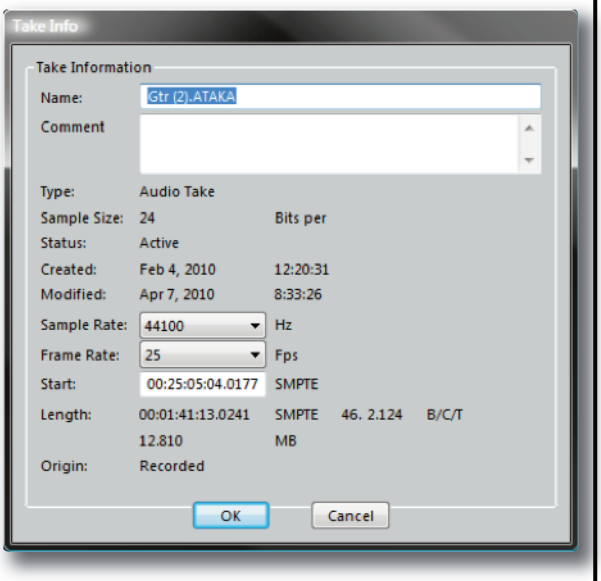



**Conversion Options:**
Depending on the format of the file you have selected, various conversion options may be included in the menu:

- **Convert multi-channel WAV/BWF files into mono BWF Audio Takes** will convert multi channel (stereo or more) WAV or BWF files into separate mono BWF files and add them to the current folder. Since Soundscape only works with multiple mono files, this process is automatically done in the background when multi-channel WAV/BWF files are dragged into the Arrangement Window.
- Clicking on **Convert mono or multi-channel WAV/BWF files into mono ATAK Audio Takes** will convert mono or multi channel (stereo or more) WAV/BWF files into single/multiple mono Soundscape ATAK files which can directly used inside the Arrange Window.
- Selecting **Convert mono Audio Takes into multi-channel BWF files** will convert mono audio take file(s) (ATAK or BWF) into multichannel BWF files. This is useful for mixdowns or stems you want to use in CD Burning or DVD Authoring applications.
- Selecting **Convert mono ATAK Audio Takes into mono BWF Audio Takes** will convert proprietary SSL Soundscape ATAK files (always mono) into BWF files, for use in other Audio Applications.
- **Convert to mono BWF Audio Takes** will convert a mono file into BWF format.
- **Convert to mono ATAK Audio Takes** will convert a mono file into ATAK format.




Depending on the format of the selected audio file, a Properties entry may be displayed in the menu. This will open a dialogue box like the one shown to the right. You can **Name** the Take, add a **Comment**, view file details, and set the **Sample Rate**, **Frame Rate** and **Start** time.

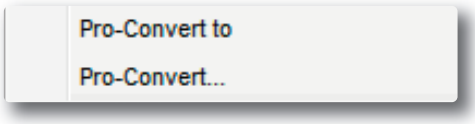


### Right clicking on an Arrangement File

Right clicking on an Arrangement file opens another menu.

**Rename**, **Cut** and **Copy** behave in the same way as you would expect in Windows Explorer.

**Delete, Delete empty Folders trees only** have been explained above.




**Pro-Convert**

The Pro-Convert options allow you to convert the selected Arrangement into a third party format, such as Pro tools or Logic. The **Pro-Convert to** option provides a quick way to do this, whereas **Pro-Convert** opens a conversion dialogue in which advanced settings can be adjusted.

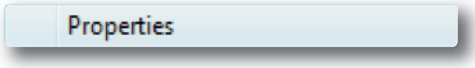
*Pro-Convert procedures are covered in detail in the 'Project Conversion' section of this chapter.*

### Right clicking on an Automation File

Right clicking on an Automation file opens another menu.

**Rename**, **Cut** and **Copy** behave in the same way as you would expect in Windows Explorer.

**Delete, Delete empty Folders trees only** and **Replace Take(s)** have been explained above.




The **Properties** option opens a dialogue box which includes 'Knobs' and 'Events' information.

## Auditioning Files and Arrangements

Auditioning (Preview, Playback) an Audio Take inside the File Manager is done as follows:

- Right-click on the Take
- Hold the Right mouse button
- Move the mouse pointer a little to the left or right

The mouse pointer will turn into the Solo tool (Ear Icon). The audio will always be played from the beginning of the Take, including any initial silence.

Takes are played back through the same track(s) as muted Parts. By default this is Track 1, but this can be changed in the **Settings** / **Muted Parts/Takes Solo Output** / **Audio** menu.

Full Arrangements can be auditioned in the same way, but they are played back through the currently active Mixer so they may not sound as expected. In particular, the active Mixer should have track inserts for all the tracks used by the Arrangement.

This is an extremely powerful function and loads a whole project in the background. Depending on the complexity it therefore may take a second until Playback of the Arrangement starts. Leading silence inside the Arrangements is naturally also 'played'.

# Opening and Dragging Files into the Arrange Window

## Opening Folders, Arrangements and Mixer Files

In the usual Windows manner, double clicking on folders opens them. Arrangements and Mixer files can also be opened by double-clicking on them in the File Manager.

## Dragging Takes and Arrangements into the Arrange window

A Take (Audio or Automation) can be dragged (click+hold mouse) from the File Manager and dropped (release mouse) at any time position on any virtual track inside the Arrange window as a Part. WAV and BWF files can also be dragged directly from Windows Explorer into the Arrangement. Dragging/dropping will take the current Snap setting into account (if active).

Takes can only be dragged and dropped one by one, even if multiple Takes are selected. Stereo and multichannel audio Takes are automatically dragged together and become Stereo/Multi Linked Parts inside the Arrangement. This requires that Takes have the same length and the same name (appended with L / R or the Multi-Ch extension).

If you drag a Take from the File Manager into the Arrange window and drop it on a virtual track that has an output assignment in the Record Track Column, then the Part will automatically be assigned to that Track and be colour-coded accordingly.

---

*WARNING: If the Sample Rate of a dragged Take is different to the Sample Rate of the Arrangement, it may be converted automatically depending on the setting of the Auto Sample Rate Convert Process Audio Takes when dragged into Arrangement option under* ***Settings*** */* ***Preferences****.*

---

The Sample Rate Convert Process tool performs the conversion, and as for all audio processing tools the source audio Take(s) may be deleted automatically, depending on the setting of the audio processing tools behaviour (**Settings** / **Preferences**)

Arrangements can be dragged into the Arrange window in the same way as Takes. This will not replace the active arrangement, but add the Parts / append an arrangement. Part and Track assignments will also be adopted automatically when record Tracks are defined on destination virtual Tracks.

## Dragging Takes or Arrangements to original Time Code using the Ctrl Key

A Take file can be dragged from the File Manager and dropped (as a Part) at the time position where it was originally recorded (ie., the start time shown in the File Manager, Time Stamp inside a BWF File), by holding down the ***Ctrl*** key while dragging/dropping.

Dragging an Arrangement to the original Time Position works in the same way.

**Important:**

- If the original time position is outside the currently displayed time range, you will not see the dropped audio.
- If the error message Parts cannot be dropped out of range appears, the original time position could be earlier than the start of this Arrangement (due to SMPTE Offset) or is outside the maximum time range that is possible for one Arrangement (approximately 13.5h)

## Dragging Takes or Arrangements to a Locator using the Alt Key

To drag a Take or Arrangement from the File Manager to a Locator position in the Arrange window, press the ***Alt*** key while dragging/dropping and the beginning or end of the Part or any Part inside the dragged Arrangement will be positioned at the nearest Locator automatically.

# PROJECT CONVERSION

## Introduction

Soundscape 7 can read and write several third party project formats directly using the File Menu or context sensitive right click menu's in the File Manager Window, thanks to the build in SSL Pro-Convert technology.

### Reading and Writing 3rd Party Audio Projects

The Pro-Convert Engine can process audio related timeline information from any supported source project file and translate this to Soundscape V7 Arrangement. The supported information contains:

Edits, Regions, standard Fade Shapes, Track Names, Media Files (Audio + single Video Track), Markers/PQ Data, Volume and Pan Automation, Region/Clip Gain and Cross-fades.

When saving an Arrangement as a 3rd Party project the same information is exported to the destination format.

### What Pro-Convert doesn't do

Midi, Instruments, REX and Apple Loops, Mixers and Plug-Ins, custom Clip Gain Envelopes, Pitch and Time-corrected Tracks, anything that can't simply be translated or isn't present in all supported DAWs, is unsupported information Pro-Convert cannot process.

All modern DAWs, however, offer a "Freeze", "Consolidate", "Bounce to Disk", "Commit to Disk" or similar function that can render not supported events into standard audio tracks, before the projects are opened with Soundscape.

### Audio File conversion

Pro-Convert can also translate the audio files for you. In various project conversions this is mandatory, when source and destination DAWs are not be able to use the same audio file types, the same or multiple bit depths or share the same strategy on how to create stereo tracks (interleaved Stereo vs. two Mono files).

| File Format | Description | Ext. | Supported Bit Depths | Supported on | Notes |
|---|---|---|---|---|---|
| Wave | The major audio format on Windows OS | *.wav | 8, 16, 24, 32 linear, 32 float | Import and Export | *Bit Depth support is Host dependant* |
| Broadcast Wave | Wave Format including Metadata and Time Stamps | *.wav | 8, 16, 24, 32 linear, 32 float | Import and Export | *Bit Depth support is Host dependant* |
| Audio Interchange File Format | The major audio format on Mac OS | *.aif | 8, 16, 24, 32 linear | Import and Export | *Bit Depth support is Host dependant* |
| Audio Interchange Format (comp.) | Adds Compression Support to AIFF files | *.aic | 8, 16, 24, 32 linear, 32 float | Import and Export | *Only uncompressed files are supported!!!* |
| SSL Soundscape Take File Format | Soundscape Audio File-format on Editor v4.2 or later | *.atak | 16, 24 | Import and Export | *Only supported when residing on Windows Drives, SDisks are not supported* |

Conversion is never going to be perfect, in order to get the most accurate (≈100%) representation of a project, Soundscape ALWAYS needs to store a project in it's own Arrangement Format first before export to another format.

On Import, ANY 3rd party project will first be saved as a Soundscape Arrangement, which is then opened.

## Successful Imports and Exports...

...largely depend on audio files being where they should be, Projects being Sample and not Tick based and Destination folders being accessible.

Also, the Project Files need to be written in a version that is known to Pro-Convert.

If you want to import a project into Soundscape, please also do remember to load or create a Mixer.

| Project Types and File Versions | Ext. | Description | Notes |
|---|---|---|---|
| **Pro-Tools 7.x to 9.x Session File**<br>Import V 7.x to Version 9.x<br>Export V 7.x are created | *.ptf | AVID's session format for Pro Tools Verson 7 to 9.x. Pro-Convert always exports sessions in the 7.x format. | *SDII audio Files are NOT supported.* |
| **Steinberg 3.x to 5.x XML Track archive**<br>Import V 3.x to Version 5.x<br>Export V 3.x are created | *.xml | XML format that can be used with Steinberg Nuendo 3.x and later and Cubase SX3.x and later. | - |
| **Samplitude EDL**<br>Import V 10.x<br>Export V 10.x<br>(also supports Reaper and Sequioa) | *.edl | .edl format generated by Magix Samplitude/Sequioa or Cockos Reaper. | |
| **Apple Final Cut V1-2.x XML**<br>Import V 1 to Version 2.x<br>Export V 2.x are created | *.xml | XML format that is used with Final Cut Pro 4 and later, Logic 8 and later and Soundtrack Pro 2 or later. | *Logic 8 and 9 do not export any edits, but bounce all Tracks as single files.* |
| **SSL Soundscape Arrange**<br>Import V 6.2 or later<br>Export V 6.2 or later are created | *.arr | Arrange file format; Audio Project Format used by SSL's Soundscape™ V6.2 or later. | |
| **SSL Soundscape Editor Arrange**<br>Import V 4.x to Version 5.5<br>Export V 5.5 are created | *.arr | Arrange file format; Audio Project Format used by SSL's Soundscape™ Editor v4.x or later. | *Files on SDisks are not supported* |

*NOTE: Tick-based ppq resolutions differ between Sequencers; In order to maintain project timings during conversion, Pro-Convert does NOT support Audio projects with Tick-based time.*

## Installation

Pro-Convert installation is described in the installation section towards the start of the manual.

*NOTE: All described Pro-Convert menu entries are only available if Pro-Convert for Soundscape has been installed, and a valid Soundscape password has been entered in Soundscape's **Options** dialogue.*

## Opening Third-Party Projects

To open third-party projects, select **File** / **Open Arr...**, browse to the desired folder, and change the visibility option (in the bottom right-hand corner of the window) to the file type you want to open (or **All Files**).

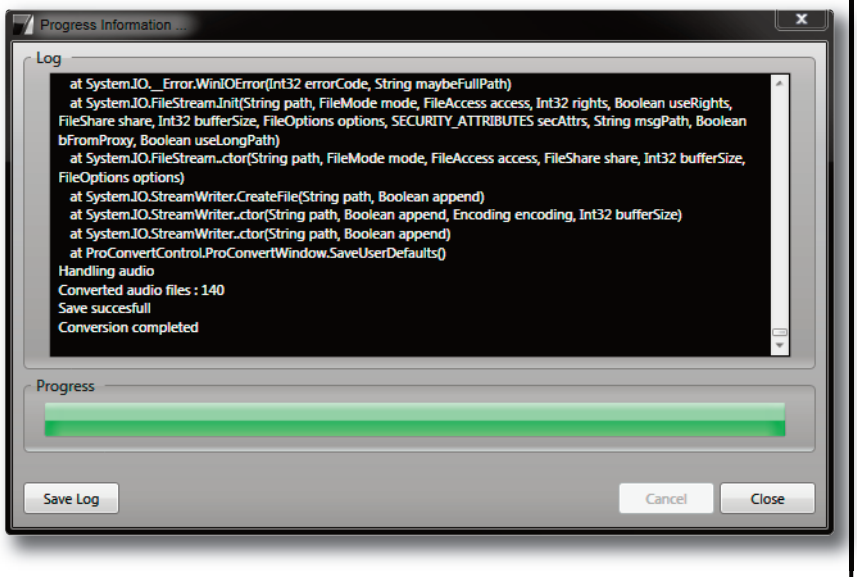
As an alternative you can simply double click on any recognised third-party project inside the File Manager window.

Pro-Convert for Soundscape translates audio track related information in the third-party project file to a Soundscape Arrangement, using default conversion settings, and if required, also convert the audio files. At the end of the process a Soundscape Arrange File is written to the same folder and automatically opened.

A progress and logging window will appear during the conversion, similar to that shown below.

## Quick-Saving in Third-Party Project Formats

To save a session in a third-party project format, select **File** / **Save ARR and Pro-Convert to...** and browse to the desired folder. Define the name and file type, and press **Save**. This will first save the currently active Arrangement and then create a new file in the selected third-party project format with standard conversion settings in the same folder.



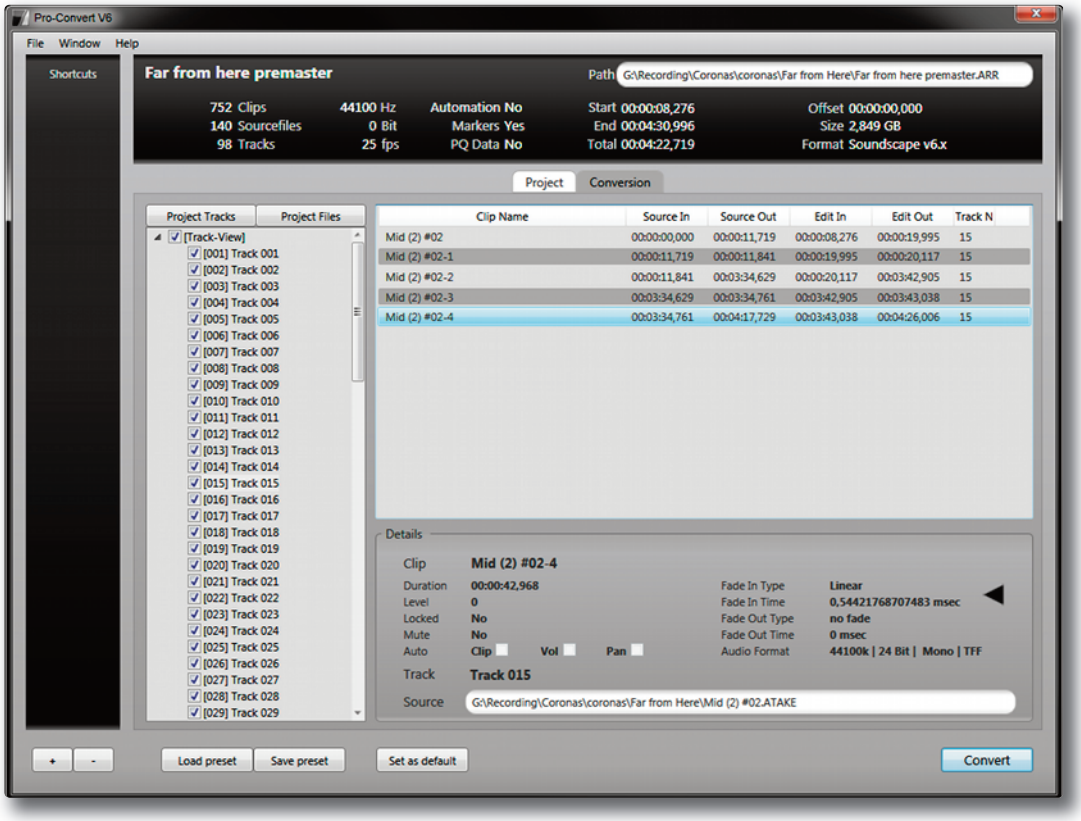

Alternatively, if the file you wish to convert is not open, right-click on any supported Project in the File Manager, select **Pro-Convert to** from the menu which appears, and select the format in the subsequent sub-menu. This will create a new third-party project file in the same folder.

With either method, a progress and logging window will be displayed during the conversion, similar to that shown above.

*NOTE: For best results, a copy of older Soundscape Arrangements (Arr. Created with V6.2 or earlier) should be saved as Version 7.*

## Full Pro-Convert Window

When saving a session in a third-party format, it is possible to define a wider range of options. To do this, select **File** / **Save ARR and Pro-Convert** (or right-click on any supported Project in the File Manager, and select **Pro-Convert** from the menu which appears). The full **Pro-Convert** window will open:




The interface is streamlined to allow quick conversion for standard studio tasks and recurring jobs, while also allowing you to delve deeper into the project structure and alter the project output prior conversion. Project information for the open file is displayed across the top of the window.

The **Convert** button starts the conversion to a target project. The button is only active when all target options are set up properly.

*Note that the Arr file will also be saved when **Save ARR and Pro-Convert** is selected.*

*The **Shortcuts** area and **Load/Save Preset buttons** to not have any function inside Soundscape.*

## Project Tab

When the **Project** tab is active (with its label displayed in white) information about the opened Source project is displayed.

This section is divided into three main areas:

- Project Browser on the left;
- Clip Browser to its right;
- Clip Details, shown below the Clip Browser.

### 1. Project Browser

The two buttons at the top of the project Browser define whether the rest of the window displays all the audio tracks (**Project Tracks** selected) or all the source files (**Project Files** selected) used inside the project.

When in **Project Tracks** View, you can exclude full tracks from the conversion by un-ticking their check box. This can be useful to create different target projects excluding Dialogues or Mix downs or to create different language versions of a project.

### 2. Clip Browser

The Clip Browser shows a list of all clips (or edits) on a selected Track (**Project Tracks** active) or for a selected source file (**Project Files** active). The **Clip** (Part) **Name**, **Source** (Take) **In** and **Out** times, **Edit** (Timeline Position) **In** and **Out** times, and **Track No** (numbers) are displayed.

Source I/O times refer to the sector that is used from the original audio source; Edit I/O times refer to the start and end of the Clip inside the timeline.

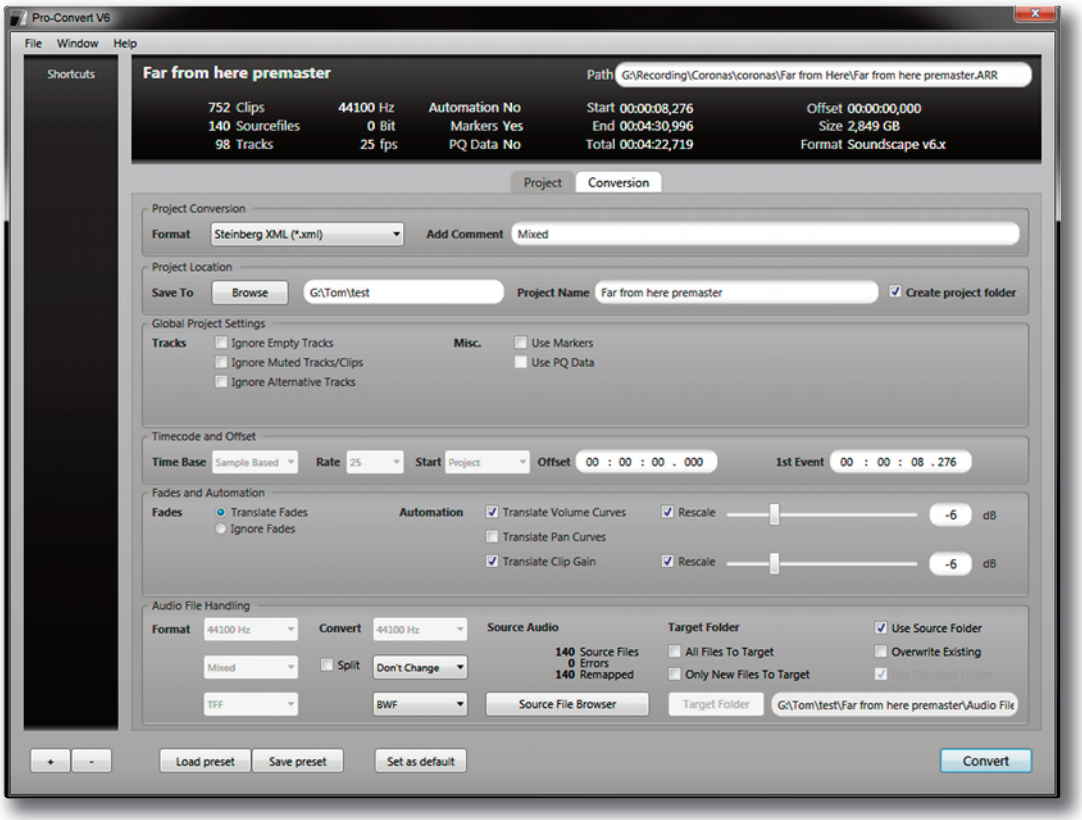
Clicking on any of the Clips reveals more information in the **Details** area below the Clip Browser.

### 3. Clip Details

Clips in Pro-Convert are the smallest piece of edit information inside the timeline. The following details are extracted from the source project: Clip **Name**, **Duration**; Gain (**Level**); Time **Locked** status; **Mute** status; **Auto**mation present for **Clip** Gain, **Vol**ume or **Pan**; **Fade In/Out Type** and **Time** (can also relate to Cross-Fades), **Audio Format**, **Track** number in the project, and the referenced audio **Source** path.

## Conversion Tab

All settings for the conversion to the Target Project are altered inside the **Conversion** tab.



The Conversion setup is divided into six logical sections, explained over the following pages:

### 1. Project Conversion:

Here you can select the target **Format** and add a comment for the conversion

### 2. Project Location:

Here you can select the disk and sub folder to which you want to save the new project and name the new file, using standard windows file navigation procedures. If you would like a project folder to be created for the new format, select the **Create project folder** check box.

### 3. Global Project Settings:

This section shows format-specific settings.

The **Tracks** options allow you to ignore empty tracks, muted Tracks and clips, and alternative takes/sub-tracks.

If is present in the source project and possible in the target format, Markers and PQ-Data can be selected for conversion in the **Misc** area.

If the Source Project contains one or more Video references, the **Video** area will be displayed on the right. One Video can be included into the conversion process.

> ***IMPORTANT:*** *Pro-Convert does not convert or trans-code at all. It simply copies the selected original video file to the new target destination project folder (if this option is selected) and adds a reference inside the target project file (if possible).*

If a video-file (or multiple video files) is present, the **Video file browser** can re-map the Video file and, if the source project contains more than one Video track, select the video track that should be included in the conversion.

### 4. Timecode and Offset

This section displays the projects timecode settings and project Offset. No settings in this section can be changed.

### 5. Fades and Automation

This section controls the handling of Fades and Cross-Fades, as well as Clip Gain, Volume and Pan automation.

**Fades**: With **Translate Fades** checked, Pro-Convert translates all fades and crossfades to the target project using the best possible match for standard Fade Curves between Source and Target. Any sort of custom fade shape is not supported and should be removed from the source project, prior to using Pro-Convert.

With **Ignore Fades** selected, all Fades will be removed from the project during conversion.

**Automation**: Depending on source and target format capabilities, the following options may be available:

If **Translate Vol Curves** is checked, volume curves and volume automation will be translated. In most cases, these curves control the DAW's volume faders in the mixer. To adjust volume automation data which goes beyond the maximum possible value in the target format, all existing volume automation data can be reduced by selecting the **Rescale** box (which appears when **Translate Vol Curves** is active) and moving the slider which then appears to its right. The rescale amount is shown in dB.

If the source format contains pan automation, activating **Translate Pan Curves** allows them to be translated to the target project. In most cases, these curves control the DAW's pan pot in the mixer.

The **Translate Clip gain** option allows to use Clip Gain Information from the source project and when destination projects support this feature, create Clip Gain in the target project. To adjust gain data which goes beyond the maximum possible value in the target format, all existing gain data can be reduced by selecting the **Rescale** box (which appears when **Translate Clip gain** is active) and moving the slider which then appears to its right. The rescale amount is shown in dB.

## 6. Audio File Handling

The **Format** area displays the Sample Rate, Bit Depth and Audio File Format of the source project, while the **Convert** area displays this information for the destination project. During conversion, Pro-Convert will generate new audio files for any source audio in the bit depth and audio format selected in the **Convert** area. Destination Bit Depth and Audio File Format can be changed; Sample Rate is a display only.

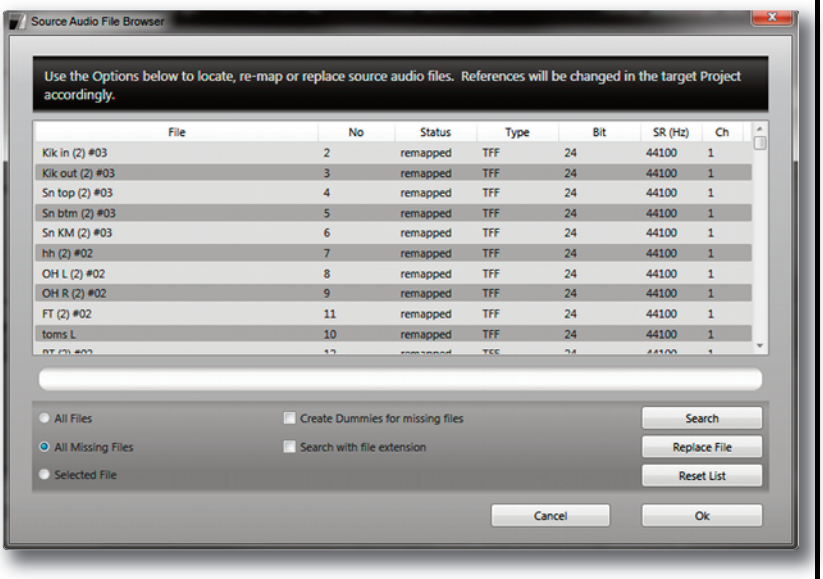
When **Split** is checked, Pro-Convert splits interleaved stereo files into two mono files, creating duplicated tracks and track automation accordingly. This option is checked automatically when converting to Pro-Tools and SSL Soundscape.

The **Source Audio** section displays the status of the Source Audio Files, (Total No. of Files, found/not found/remapped), that have automatically or manually been re-mapped (or re-linked) by the User with the Source File Browser. **Errors** are displayed when any Audio Files cannot be found.

Selecting **Source File Browser** opens a window listing the included audio files, and their status (ie. **not found** or **located**).

There are various reasons why a project contains invalid audio file pointers – for example, if Pro-Convert cannot find certain audio files in the path specified inside the source project file, an Audio file was not included in back-up/transport media, or the Project is using windows-specific paths (eg. on a MAC Volume which is now the Windows D:/ drive).

If certain entries in the list display the **not found** status, the lower area of the search section will allow you to set the search options:




**All files**: Performs a detailed search for all audio files in the source project. This option will re-map all referenced files.

**All missing files**: A search is performed for all files that are not found.

**Selected File**: Search for the file that is currently selected in the list.

**Search with extension** If this box is unchecked, a search is performed for media files that have no extension. Deactivate this option if the referenced media files have a file extension in the project but not on their disk location. (for example, if media files have been transferred from an older MAC HFS disk).

**Create dummies** If **Create dummies for missing files** is selected, a substitution file will be created for every missing file, which simply contains audio silence. The newly created file will have a **dmy** prefix added in front of the original file name.

**Search** Click on **Search** to open the browse window and select a starting directory for the search. You can use “wildcards” (eg. bass*.wav) if you are not sure about the exact name of the replacement file. You can also use a wild card for the file extension (eg. bass 06.*) to replace the referenced file with a file of the same name but different file format.

**Replace file** To select a replacement file for the currently marked file in the list, click on **Replace file** so that it ‘glows’ blue, and select the file from the browse dialogue.

**Reset List** This button resets all replacements and re-mapping, and reloads the original state.

The **Target Folder** area of the **Conversion** tab is used to define elements of the folder into which the new session will be saved:

The **Target Folder** button is used to choose or create a destination folder for audio files. If **All Files to Target** is selected, all files will be copied/converted into this folder; If **Only New Files to Target** is selected, only new audio files will be put there.

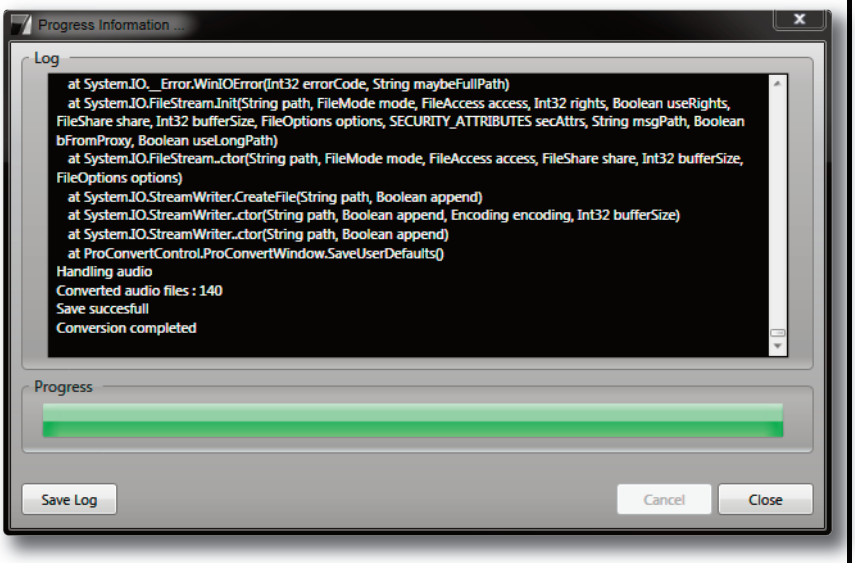
If **Use source folder** is selected, new files will be placed in the same folder as the original files. Original files are left untouched. If no audio file conversion is required (ie. source and target project can both use the same files) no file copy or conversion will be performed.

If **Overwrite existing files** is active, existing files with the same name will be overwritten in the target folder without further warning. For safety reasons, you should leave this option inactive unless you are certain of what you are doing!.

If **Use Standard Folder** is active, relevant sub folders (such as 'audio files') will be created in the project.

The **Progress Info Window** shows a progress bar during conversion, along with a logging section showing all activity of Pro-Convert since the window was last opened.

The **Save Log** button can be used to save logging information to a text file. This may help you finding the cause of problems with a particular source file.



# Project Format Reference

| | Supported Projects | Supported Data | Before you can Import into Soundscape | Before you export to another format | Known Limitations |
|---|---|---|---|---|---|
| **Apple FCP XML** (FCP V4-7, STP V2, Logic Pro V8-9) | 99 Audio Tracks, .wav / aiff audio files, Sample Rates 32 kHz - 96 kHz, 16 - 32-bit files. Sample based Projects only!!! | Single Volume and Pan Automation Curve per Track, Corssfade/Fade (lin+exp), Names and Markers | All non PCM Audio Tracks need to be bounced prior to Export (Instruments, MP3, REX etc.), only single Video files | Clip Gain is not supported in Apple XML and should be consolidated. | Logic 8 and 9 bounce all Tracks before exporting to XML, hence all Edits are lost. |
| **AVID Pro-Tools PTF** (AVID Pro-Tools V7-9) | 256 Audio Tracks (with Alternative Tracks), .wav / aiff audio files, Sample Rates 44.1 - 96 kHz, 16 or 24-bit files. Sample based Projects only!!! No SDII file support! | Fades/Crossfades, Names, Markers, Pan and Volume Automation. | Sessions created with earlier (ie. 6.9) or later Versions (ie. 10.1) need to be saved as a 7 - 9 PTF session in Pro-Tools | Pro-Tools V7 Session Format does not know about Clip Gain, so this Information will be ignored. In case of mixed bit depths all 16-bit audio files will be converted to 24-bit. | Special Characters should be avoided when converting from or to PT. Pro-Convert does not create Stereo Tracks on export (but multi mono tracks) |
| **Magix Samplitude EDL** (Samplitude/Sequoia v10 and Cockos Reaper) | 256 Audio Tracks, .wav / aiff audio files, Sample Rates 44.1 kHz - 96 kHz, 16 - 32-bit files. | Volume and Pan Automation per Track, Crossfade/Fade (standard types), Clip-Gain, Names, Markers and PQ Data | All non PCM Audio Tracks need to be bounced prior to Export (Instruments, MP3, REX etc.) | - | - |
| **Steinberg Cubase /Nuendo XML** (Cubase/Nuendo V6) | Trackarchive XML with 256 Audio Tracks, .wav / aif, 16 - 32-bit. Sample based Projects only!!! | Volume and Pan Automation per Track, Crossfade/Fade (standard types only), Clip-Gain (no custom curves), Names, Markers and PQ Data | Export "selected Tracks". All non PCM Audio Tracks need to be bounced prior to Export (Instruments, MP3, REX etc.) | Import "Trackarchive". All non PCM Audio Tracks need to be bounced prior to Export (Instruments, MP3, REX etc.) | Special Characters should be avoided. |
| **SSL Soundscape V6-7 ARR** | 1024 virtual Tracks, .wav / atak (Take File Format), Sample Rate 44.1 - 96 kHz, 16 and 24-bit files | Volume and Pan Automation per Track, Crossfade/Fade, Clip-Gain, Names, Markers and PQ Data* | Arr files created with SS 6.0 or 6.1 should be saved with SSL Soundscape V7 before being converted with Pro-Convert. | Soundscape's max level for volume automation data is +6dB, the max. level for clip is 0dB, this may require rescaling. | Soundscape does not support 32-bit Audio Files. Hence all 32-bit files will be converted to 24-bit PCM. |
| **SSL Soundscape Editor 4-5.x ARR** | 256 virtual Tracks, atak (Take File Format), Sample Rate 44.1 - 96 kHz, 16 and 24-bit | Volume and Pan Automation per Track, Crossfade/Fade, Clip-Gain, Names, Markers and PQ Data * *(See below)* | Arr and Audio Files need to reside on Windows Disks. Projects should be saved with V7 prior conversion to any other format. | Soundscape's max level for volume automation data is +6dB, the max. level for clip is 0dB, this may require rescaling. | Soundscape SDIsks are not supported, files need to be copied to Windows drives. |

* *100-199 Begin of Track*
*200-299 End of Track*
*300-999 Subtrack/Index*

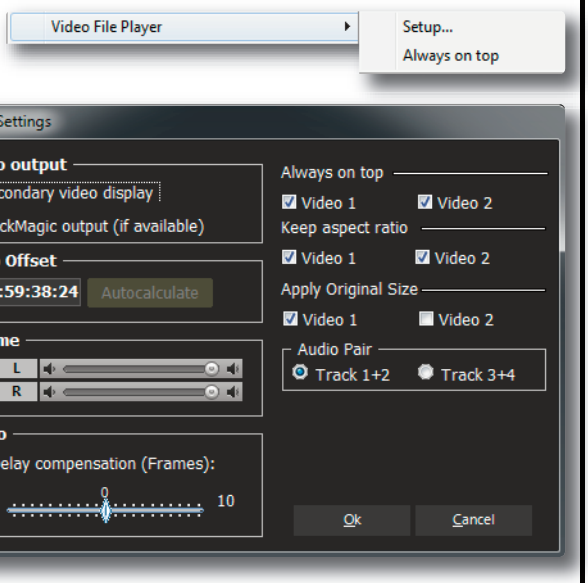
# Video File Player Window

Soundscape has a built-in video player which can play video files synchronised to the Soundscape timeline. Up to two video file displays can be shown, allowing for studio floor monitoring of the video footage, for example. You can open Video Files via the **File** / **Open Video File** menu.

To open or hide the primary video window, toggle the **View** / **Video Player** option, or press the ***V*** key on your keyboard.

## Video Settings

Clicking **Settings** / **Video File Player** / **Setup...** opens the **VideoCue Settings** dialogue.




In the top-left of the display, there are two **Video Output** options: Click on the **Secondary video display** check box to open a second window for the video file. If you have a BlackMagic video card, activate it by clicking on the **BlackMagic output (if available)** check box.

It is possible to introduce a timecode offset for the video file, to match Soundscapes timecode to that which is displayed within the video. To set this offset, start by playing a few seconds into the file (you may need to close the **VideoCue Settings** window to access the transport!). Now re-open the **VideoCue Settings** window, type the timecode displayed in the video into the **Auto Offset** field and click **Autocalculate**. The number you typed will change to show the offset value.

To adjust the level of the video's **L**eft and **R**ight audio channels, use the **Volume** sliders below the **Auto Offset** area. If the audio and video are out of alignment due to a delay introduced within the system, you can compensate for it using the **Delay compensation (frames)** slider at the base of the window – this advances or delays the video file's audio by up to ten frames.

Below that, there are three options which can be set individually for **Video 1** and **Video 2** (the secondary display): Each video window can be set to be **Always on top**, to **Keep aspect ratio** of the video file, and to always **Apply Original Size**.

*If **Always on top** is selected, the Video File Player window will always remain visible on top of any other window, even if that other window is active (except if that other window is also set to be Always on top. In Soundscape this could be the Mixer or Big Current Time window).*

*Note that Video 1 can also be set to Always on top using the **Settings** / **Video File Player** / **Always on top** option*

Finally, you can define which pair of the video file's audio tracks to use, in the **Audio Pair** area of the window: click on either **Track 1+2** or **Track 3+4**.

Click **Ok** to close the window and action any changes, or **Cancel** to close the window and cancel any changes within it.

### Supported Video File Formats:

The following file types and formats are supported in Soundscape 7:

- AVI, WMV, QuickTime (including Pro Codecs), DIVX, h264 (MP4/M4V) and MPEG.
- Frame sizes up to 1080 (HD 1920 x 1080) are supported, as are all frame rates (including 23.98, 50 and 60).

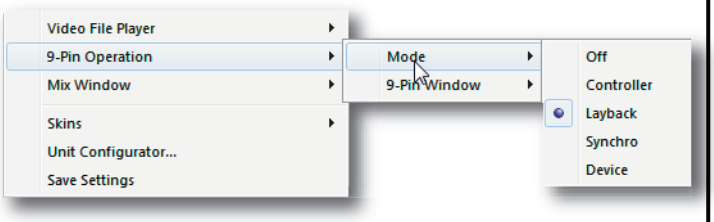
# 9-Pin Window (Soundscape 32 Systems only)

## Overview

This feature is only available when a Soundscape 32 equipped with the Sync Option is included as Unit 1 in the current Unit configuration. The 9-Pin Control RS-422 port at the back of the Soundscape 32 unit (labelled **Serial**) is used to connect to a 9-Pin device.

The **9-Pin Operation** item in the **Settings** menu will appear solid if 9-Pin operation is available. It will otherwise be greyed out.

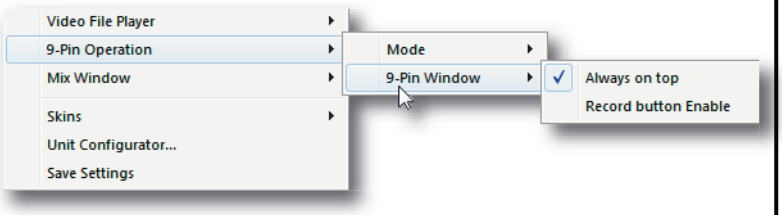
The **9-Pin Operation** / **Mode** submenu lets you choose between **Off** status and four possible 9-Pin Modes (see overleaf).




The required mode can also be selected by clicking the **Mode** box in the upper left corner of the 9-Pin Master window.

The current mode is saved when **Settings** / **Save Settings** is clicked.

Clicking on **9-Pin Window** in the submenu lets you activate or deactivate **Always on top**.




If active, the 9-Pin Master window will always remain visible on top of any other window, even if that other window is active (except if the other window is the Mixer, Big Current Time, or Video File Player window and is also set to be **Always on top**).

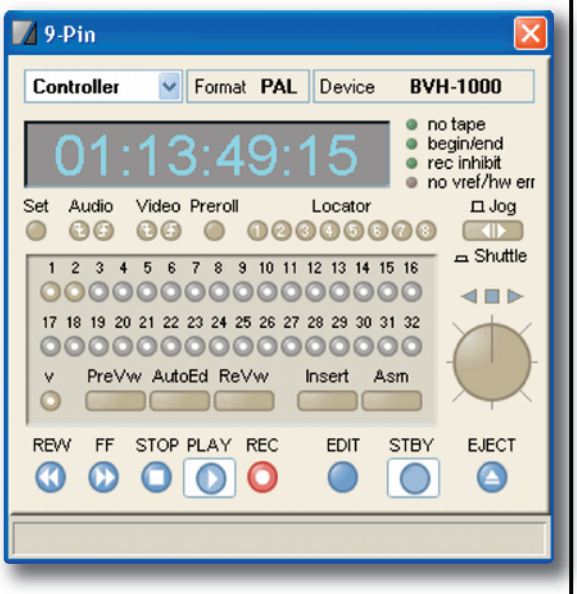
If the **Record button Enable** option is selected, the Record button in the 9-Pin Master window can be used in the Controller, Synchro and Layback modes to put the 9-Pin device into Record mode. If it is not selected the Record button appears dimmed and clicking on it has no effect.

The 9-Pin Master window can be shown or hidden using the corresponding entry under the View menu or the key command ***U***.

## 9-Pin Modes

### Controller mode

In this mode, the 9-Pin Master window acts as an independent remote control unit for the connected 9-Pin device. The controls operate as explained below.




#### Set

If **Set** is activated, the Current Locator position will be stored in the corresponding function memory of the device when the **Audio In Pt**, **Audio Out Pt**, **Video In Pt**, **Video Out Pt**, **Preroll**, or **Locator** button is pressed.

#### Audio In Pt, Audio Out Pt, Video In Pt, Video Out Pt

If **Set** is activated, clicking one of these buttons stores the Current Locator position in the corresponding 'In' or 'Out' point memory of the 9-Pin device.

If **Set** is not activated, clicking one of these buttons cues the 9-Pin device to the corresponding 'In' or 'Out' point.

**Preroll**
If **Set** is activated, clicking this button stores the 'seconds:frames' portion of the Current Locator position in the **Preroll** memory of the device.

If **Set** is not activated, clicking this button cues the device to the **Preroll** position.

**Locator 1-8**
If **Set** is activated, clicking one of these buttons stores the Current Locator position in the corresponding **Locator** memory.

If **Set** is not activated, clicking one of these buttons cues the 9-Pin device to the corresponding **Locator** point.

**Audio track arming 1-32**
These buttons control the track arming status of the 9-Pin device's corresponding audio tracks. By default, only the first two analogue audio tracks can be armed.

You can specify the number of analogue or digital audio tracks for each device by adding a device specific line to Soundscape's .INI file. See below for more details.




**Video track arming**
The **V** button controls the track arming status of the 9-Pin device's video track.

**Preview, Auto Edit, Review, Insert, Assemble, Edit, Standby, Eject**
These buttons control the corresponding functions of the device.

**Rewind, Fast Forward, Stop, Play, Record**
These buttons control the corresponding tape transport functions of the 9-Pin device.

The **REC** button can only be used if the **Record button Enable** option in **Settings** / **9-Pin Operation** / **9-Pin Window** is selected. If it is not selected the **REC** button appears dimmed, and clicking it has no effect.

**Jog/Shuttle**
Clicking this button toggles the mode of the 'Jog/Shuttle' wheel.

## Layback mode

In this mode, in addition to all the functionality of the Controller mode, the 9-Pin device is also controlled by certain Soundscape operations:




**Tape transport commands**
All Soundscape tape transport commands are simultaneously sent to the device.

In case of Rewind and Forward commands, Soundscape actually slaves the Soundscape 32 unit to the 9-Pin device, in order to keep both in sync during wind operations. Successive Rewind/Forward button presses will result in faster/slower shuttle/wind speeds.

However, in the case of Play and Record commands, in order to synchronize Soundscape 32 properly to the 9-Pin device, the SMPTE position of the device must be fed to the MTC input of Soundscape 32 using an external SMPTE/MTC converter, and Soundscape 32 must be set to MTC Chase Slave or MTC Trigger Slave mode, depending on the Master Clock source.

**Current Locator jumps**
All Current Locator jumps made in Soundscape are simultaneously sent to the 9-Pin device, so that synchronisation is maintained.

**Scrub commands**

All speed-based scrub commands (including the scrub wheel functions from external control surfaces via Console Manager) will send the 'Jog' command to the 9-Pin device, and cause the Soundscape unit to slave to the device.

All Locator based scrub commands will cue up the device in order to keep both in sync.

**Record Punch modes**

The 9-Pin device will respond in a similar way to the Soundscape 32 system for all punch modes. This means that Soundscape's Left and Right Locators, Auto Punch In and Auto Punch Out buttons also determine the way the device records.

This is achieved by sending the 'Edit On/Off' commands at the appropriate time to the device.

Soundscape can compensate for the Edit trigger delay of the device in this case. By default, this delay is set to 0 frames. You can specify the number of delay in frames for each device by adding a device specific line to Soundscape's ini file, as described at the end of this 9-Pin section.

## Synchro mode

In this mode, in addition to all the functionality of the 'Layback' mode, the device is also controlled by additional Soundscape operations:




Audio track arming commands for Soundscape 32 will simultaneously arm the corresponding audio tracks of the 9-Pin device.

In both Layback and Synchro modes, Soundscape polls and adopts to the status of the device.

This means that the local control buttons of any attached device or the buttons in the 9-pin Master Window can also simultaneously control the 9-Pin device and Soundscape for certain operations: transport control, track arming (only in Synchro mode).

Other locally controlled device operations (eg., jog/shuttle) will cause Soundscape to follow the device's Current Locator position in order to keep both Current Locators in sync.

## Device mode

In this mode, a 9-pin controller will see Soundscape as a 9-pin-compatible audio device.




All major commands are implemented, eg., transport commands, audio track arming, edit on/off, locate, jog/shuttle, A IN RESET, A OUT RESET, A IN RECALL, A OUT RECALL etc.

If **Video Sync** is selected under **Settings** / **Master Clock**, any received 9-Pin Play/Record start commands will cause Soundscape to start in sync with the next received video frame.

# Preparation: Modifying the .INI file for 9-Pin operation

## Master modes

For every device model, a line should be added to Soundscapes INI file under the [9PinWnd] section. The INI file is located at: **C:\Soundscape\ssEditorDef.ini** and can be opened with a standard Windows text editor. This line must contain device specific information that is used by Soundscape:

Currently the following three values need to be specified per device attached:

- Number of analogue audio tracks (0...4).
- Number of digital audio tracks (0...32).

*NOTE: A device cannot have both analogue and digital audio tracks, so one of the values must be zero.*

- Edit trigger delay, expressed in frames (0...99).

For example, if you are using a Sony BVW-75 that has 4 analogue audio tracks, and its Edit trigger delay is set to 3 frames, this line must be added:

BVW-75=4 0 3

Please note that this line must be added under the [9PinWnd] section, and the device model name must exactly match the device name displayed in the 9-Pin Master Window.

If the [9PinWnd] section does not exist in the INI file yet, then a Save Settings operation must be executed first .

## Device mode

As a default, the 9-pin device mode uses 'FxDE' as the Soundscape device ID. A different Soundscape device ID can be set (eg., '5178') by adding the following line to the INI file under the [9PinWnd] section:

S9PinDevID=0x5178

*Please note that this line must be added under the [9PinWnd] section.*

If the [9PinWnd] section does not exist in the INI file yet, then a Save Settings operation must be executed first .

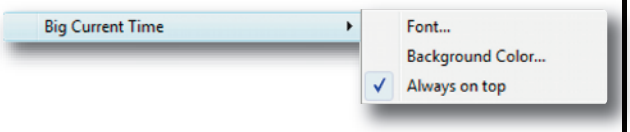
# Big Current Time Window

The **Big Current Time** window provides an additional time display that can be customised and resized. It displays the time position of the current locator either in musical time (B/C/T) or SMPTE Timecode, depending on which display has been selected for the **Time Axis** (**Settings** menu).




It can be shown or by hidden selecting/deselecting **Big Current Time** in the **View** menu or by pressing the ***T*** key.

Clicking **Big Current Time** in the **Settings** menu opens a submenu with 3 options to alter its appearance and behaviour.

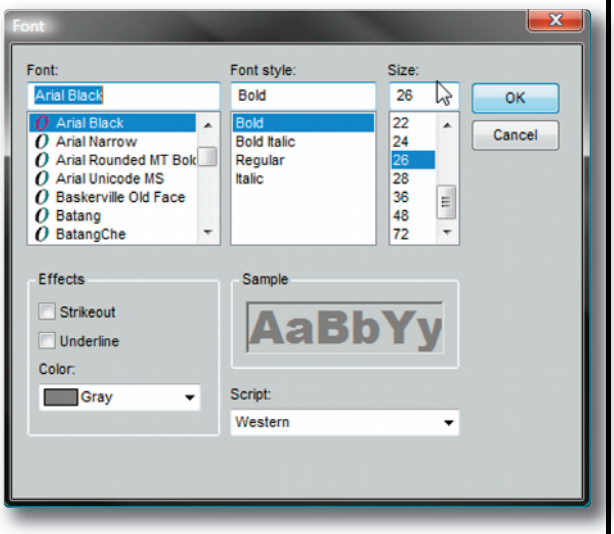



## Font

Clicking on **Font** opens a standard Windows font selection dialogue, where every installed font can be selected, and the font colour and size can be defined.




You can freely resize the **Big Current Time** Window. If you select a font face and size that wouldn't fit into the window, the time display will resize automatically.




## Background Colour

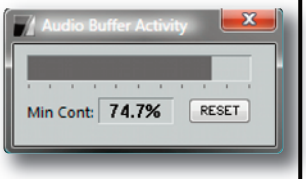
Clicking on **Background Color** opens a standard Windows colour palette to select a colour from a range of presets, edit an existing colour or create a new one.

## Always on Top

Selecting the **Always on Top** option causes the **Big Current Time** window to always remain visible on top of any other window, even if that other window is active (except if that other window is also set to be **Always on top**. In Soundscape this could be the Mixer or Video File Player window).

# Audio Buffer Activity Window

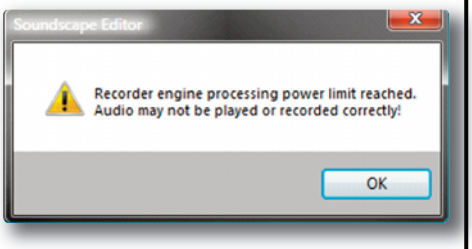
The **Audio Buffer Activity** window can be shown or hidden by pressing the ***B*** key on the computer keyboard or by checking/unchecking Audio Buffer Activity in the View menu.




## What is a buffer?

The main purpose of Soundscape is to transport audio data from sound to disk and from disk to sound, and in between to do some processing with it. As explained earlier, digital systems always need to keep a certain amount of data inside a temporary storage, in order to make sure there is always enough data and the stream doesn't stop. Also, the computer cannot dedicate all its time and attention to us, so again we need to make sure to collect audio data together before we process it.

If the Buffer Activity drops to 0% you will get the error pop-up shown to the right.



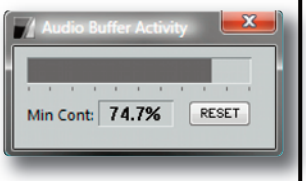

## My Buffer Activity is constantly above 80%!

Congratulations, this is a good thing! Unlike other systems, Soundscape tells you how healthy your buffer infrastructure is and not what it thinks the current taxation on the host computer is (which an application cannot reliably measure).

## What is measured?

Inside Soundscape's audio engine, each Track is part of a mini recorder/player, each of which is connected to a DSP on the audio card and to a hard disk. Any of those mini recorder/player can be located by Windows to a different CPU Core. 128 of them could work at the same time, so there is some synchronisation and Master Infrastructure that controls and connects to them all.

All of these connections need to be buffered in order to guarantee a seamless flow of data, and these are what is measured by the **Buffer Activity** Window. The grey bar shows the current buffer activity (or health). The displayed figure shows the lowest percentage that has been detected for all audio track buffers. Pressing the **RESET** button resets the readout and bar display to 100%.




## What affects buffer health?

Although it has proven to be very difficult to achieve a buffer-underrun on a modern computer with Soundscape, if the workload of the system becomes high enough and any of the track buffers is not refilled quickly enough, the recorder engine may stop.

The following variables may affect the buffer health:

**Arrangement Structure**

- The number of simultaneous crossfades and/or real-time fades
- The selected Sample Rate (higher is tougher)
- The number of active tracks being played simultaneously

**Disk Performance**

- The number of small Parts in the Arrangement (affects the Seek Times of a Disk)
- The position of the audio Takes on the disk
- The overall speed of the disk (access time, average data rate, seek times, native command cueing, RPM, On-Disk Buffers)

**Disk Fragmentation**

- Background PC Tasks
- Too many Tasks 'distract' the PC from 'SoundScaping'
- Not enough RAM available, so the PC starts to use 'Page File Virtual Memory' (=Disk)

**General PC Configuration**

- Mixpander or MX4 Card needs to share an Interrupt (IRQ) with another or multiple other PC internal Devices

- Other PC Cards are grabbing too much Bandwidth from the PCI/PCIe Bus or blocking the Bus (DMA/IRQ) for too long
- System is not configured properly, has Viruses, or Chipset/Motherboard Architecture is not good enough for Realtime Operation

### What you can do to avoid Buffer underuns...

- Raise the **Native Mixer Elements Sample Buffer Size** (in the **Settings** / **Preferences** / **Mixer** menu) to increase the size of the buffers;
- Reduce the amount of random seeking the disk has to perform by using the Mixdown or Consolidate Process tools to generate new, consecutive Takes;
- Reduce the number of real-time fades by consolidating to a new Take which includes those fades;
- Use a lower sample rate;
- Try another PCI/PCIe Slot for the Mixpander or MX4 Card;
- Remove other cards from the system and check if the behaviour improves;
- Replace the disk with a faster one, especially with bigger onboard buffers and technologies like NCQ (native command cueing);
- Replace the whole PC and get one with more CPU cores and a faster memory subsystem, try to buy a known good system configuration including components from well known quality brands.

# 5. Menu Reference

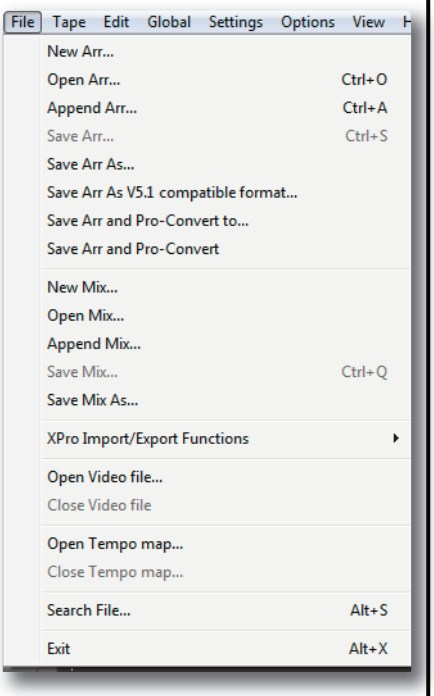
This chapter describes every function and feature in Soundscape in the order they appear inside the main menu structure. Some menus and entries are only accessible when Soundscape is in Stop Mode. Whenever a menu is not accessible during Play/Record or certain entries are greyed out/dimmed – please Stop Soundscape first.

Many menu entries are self explanatory and follow a standard Windows nomenclature, while others might need some explanation and background information before they can be used intuitively. It might therefore be a good idea to work your way through this chapter at some point (probably on a rainy Saturday) with the system switched on.

The File Manager window relies completely on 'context sensitive' mouse interaction and many features of the File Manager cannot be accessed through menus.

## File Menu

With the **File** Menu, Arrangements and Mixers can be created, loaded and saved. An Arrangement can also be appended to the current one.




There are also options to import or export to MP3, Pro Tools, SDR and OMF formats, and to open or close Video files and tempo maps.

Finally, Soundscape can be closed from this menu.

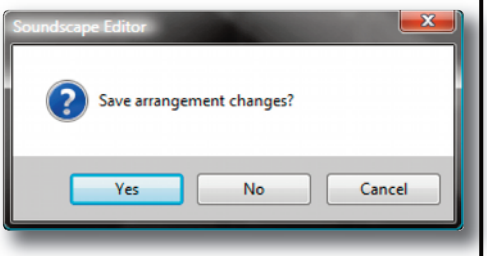
The **File** Menu cannot be accessed while Soundscape is in Play or Record Mode.

*NOTE: For the MP3 import/export to be active, the required Fraunhofer MP3 codecs needs to be installed.*
*The OMF export option is an optional plug-in.*

### Arrangements

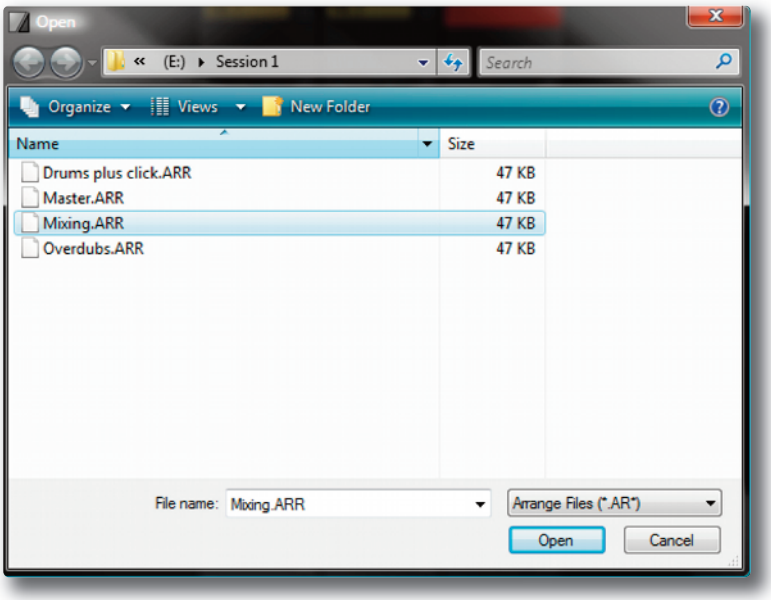



Clicking **New Arr...** will create a new, empty Arrangement window. If an Arrangement that has been edited is already active, the dialogue box shown right will appear, giving you the option to save it before closure.

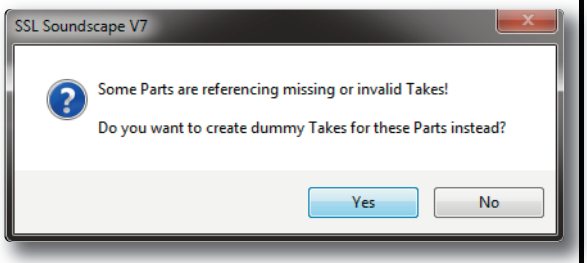





**Open Arr...** opens a standard Windows dialogue box. If an Arrangement is already active and has been edited since it was last saved, the same dialogue box mentioned above will appear.




If any Takes used in the Arrangement are not present on the Domain Root Folders, a pop-up will appear, asking whether or not you want Soundscape to create dummy Takes for those Parts. Click **Yes** to create dummy parts, or **No** to leave those Parts unloaded. Another pop-up will then appear with further explanation about your choice – click **OK** to continue.




If you have backed up these Takes to another drive it is advisable to import them to the relevant folder, then open the Arrangement again. Otherwise, the Arrangement will still play correctly except for the missing Parts.

***Important Note:*** *If you save an Arrangement with dummy Takes or missing Takes, the links to the original Takes will be lost.*

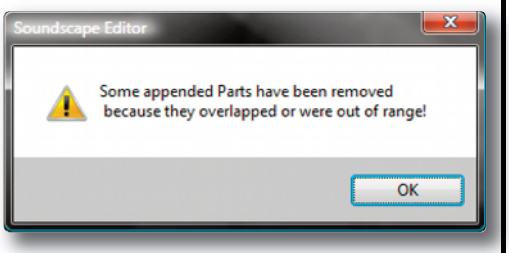



**Append Arr...** opens a Standard Windows File Browser to select an Arrangement to be appended to the active one.

The chosen Arrangement will be appended at the Current Locator position, so the Current Locator should be placed at the desired song position before using this option. If any Takes used in the appended Arrangement are not present on the relevant folder, the warning described above will be displayed.

In addition, if Part overlaps are not allowed (**Settings** / **Preferences** / **Arrangement** / **Overlapping Parts (for new edits)**), any Parts in the appended Arrangement which would overlap Parts in the current Arrangement will not be loaded.

Parts which would be out of range are not loaded either. In such cases the warning shown right will be displayed.




All the parameters saved within an Arrangement file will be loaded from the appended Arrangement, including Sample Rate, Time Code format, SMPTE offset, Varispeed setting, Record setup, Automation setup, Tempo, and Time Signature. Therefore, if you wish to retain the settings of the current Arrangement, save it, load the other Arrangement first and then append the current one.

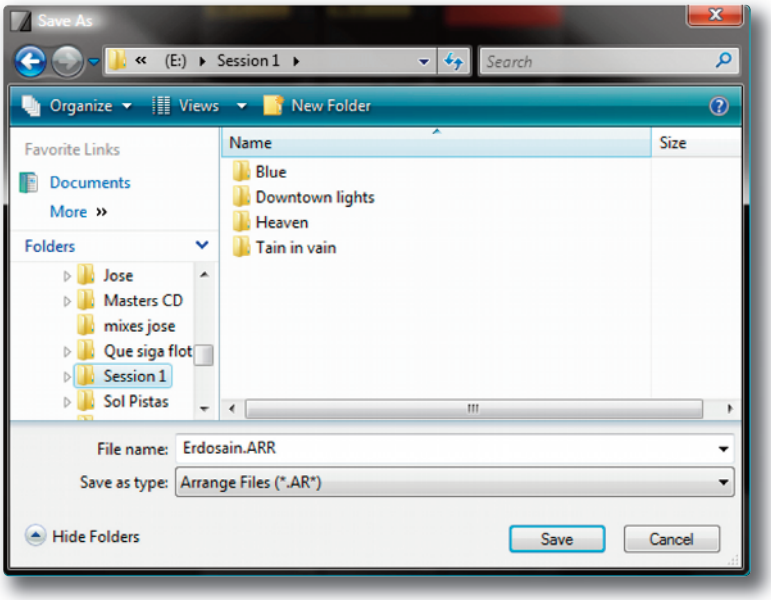
To make sure that overlaps do not occur when appending an Arrangement if they are not allowed, it is sometimes necessary to move all Parts of the current Arrangement to empty virtual tracks, while not altering the timing of any of these Parts. This can be done easily with the **Move Vertical** tool.



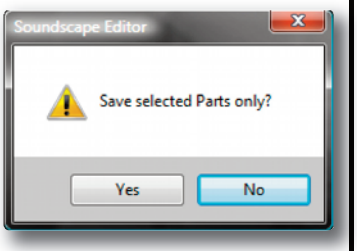

With **Save Arr...** the current Arrangement is saved and the stored Version overwritten. Arrangements include Project Information like Sample Rate, Frame Rate, Tempo, SMPTE offset, Varispeed, L and R locators, Markers, Punch and Loop button status, Settings inside Record setup / Automation setup and the assigned Record/Process folder.




Use **Save Arr as...** to save a copy of the Arrangement or to save a new Arrangement for the first time. A standard file Browser will open:

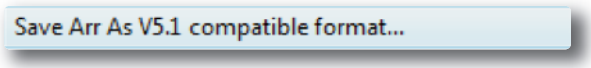



## Saving selected Parts as an Arrangement




If any Part is selected when you **Save Arr as...**, a dialogue box will appear, shown right. If you click **No**, the whole Arrangement will be saved, as described above. If you click **Yes**, only the selected Parts will be saved, with their individual parameters such as time position and virtual track position in the Arrangement, and track assignment, but without all the usual Arrangement parameters listed above.

This feature is especially useful if you wish to save Parts with an embedded snap point, quickly save a version of the current Arrangement including the muted Parts or the active Parts only, by typing ***Shift+M*** (Select Muted Parts) or ***Shift+M*** followed by ***Shift+X*** (Invert Part selections) or simply to save a Multi Track Jingle, Opener or Sound FX you need to re-use in future Arrangements.




In order to keep compatibility with legacy SSL Soundscape Editor systems, you can **Save Arr As V5.1 compatible format**. This option is only available when no WAV/BWF Takes are referenced by any Parts inside the Arrangement (in other words only the Soundscape .ATAK Format is used).

## Pro-Convert Export





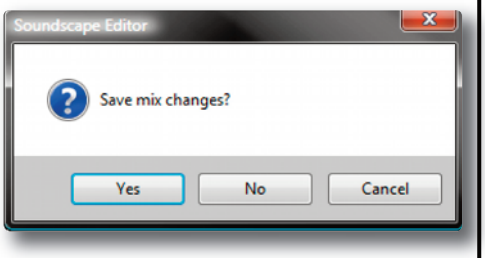


It is possible to import and export a variety of DAW formats to Arrangements. **Save Arr and Pro-Convert to** provides a quick-save function, and **Save Arr and Pro-Convert** provides a number of export options. These functions are described in full in the *Project Conversion* section of this guide.

# Mixers



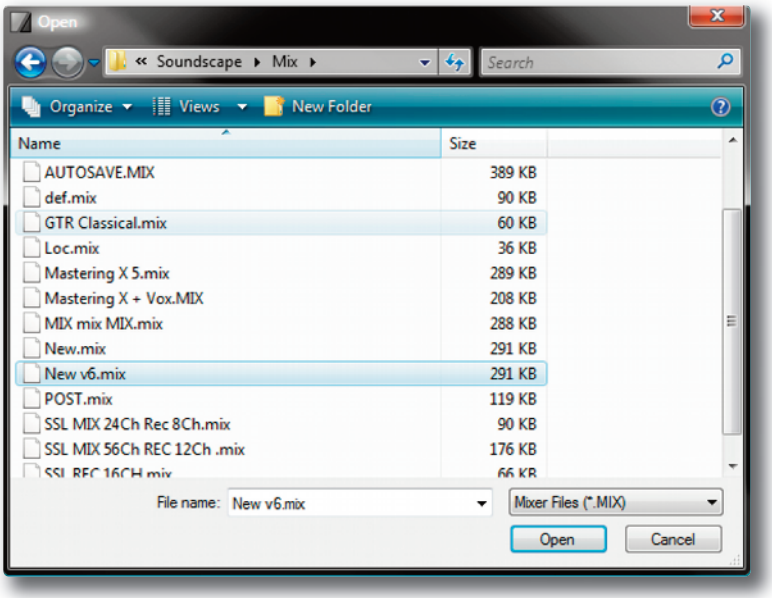

Clicking **New Mix** will create a new, empty Mixer. If a Mixer is already active and has been edited since the last save, the dialogue box shown right will appear.







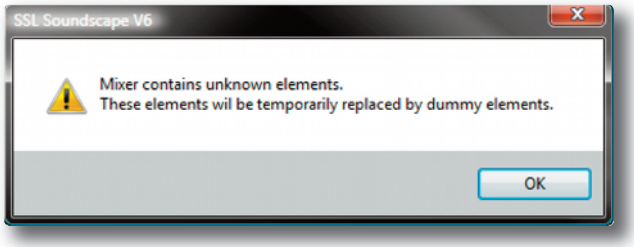
**Open Mix** opens a standard Windows file browser to search and load an SSL Mix file from any logical PC drive or network location:




If a changed Mixer is already active when you open a new Mixer, you will first be asked to save your current Mixer.

## Opening a Mixer: potential issues

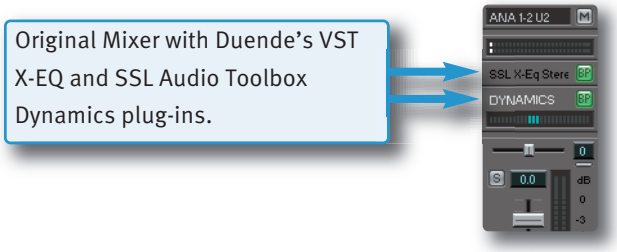
Loading a Mixer on the system where it was created is straightforward. If you attempt to load Mixers on a different SSL system containing different hardware or software configurations (eg., different number of hardware units or plug-ins that are not available).



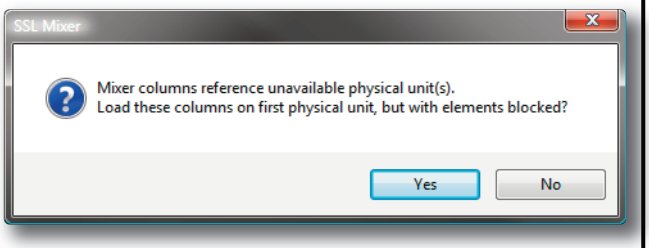

When the Mixer you are opening includes one or more mixer elements not available on the system, the warning shown left will be displayed.

The unknown mixer elements will be replaced by dummy mixer elements, giving you a chance to locate and correct the problem later. In the example below, the Mixer was created on a system using the SSL Duende's VST X-EQ and SSL Audio Toolbox Dynamics plug-ins.

When the same Mixer was loaded onto a system where these plug-ins are not available, the Mixer shows the missing plug-ins as red dummy elements:

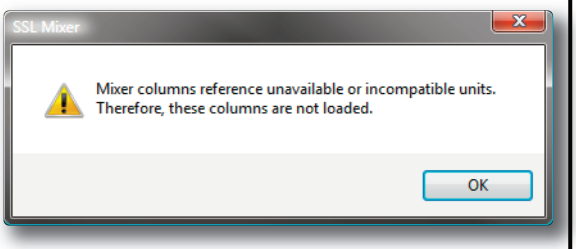


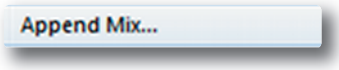

If the Mixer you are opening contains columns which refer to SSL physical units (ie., hardware units) which are not available, the dialogue box shown right will let you decide to if these columns should be loaded with their elements blocked (ie., muted) or skipped.




Clicking **Yes** will load these mixer columns. The Mute tool can be used to unmute them. These columns will be assigned to the highest numbered unit within the system. This may cause mixer elements to become muted in other mixer columns.

Clicking **No** will not load these mixer columns. The warning shown right will be displayed before the Mixer is loaded:




**Append Mix...** allows you to take the Columns from another mix window and add them to the current Mixer. The new columns will simply be appended after the last column of the current Mixer.

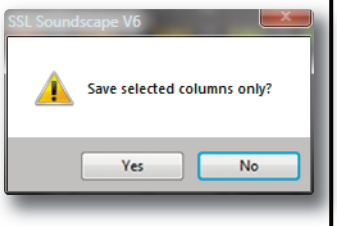



**Save Mix...** saves or updates the active Mixer. If the active Mixer has not been saved previously, a standard Windows 'Save As' browser dialogue opens.



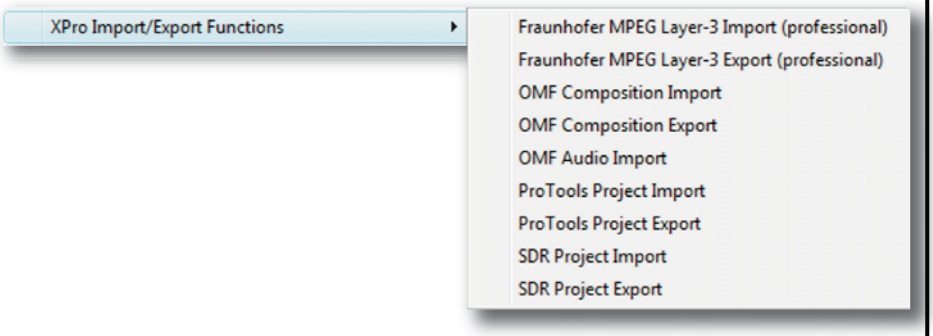

**Save Mix As...** opens a standard Windows browser dialogue. The Mixer needs to be named or renamed and you can browse your local and network drives to select the destination path.

If one ore more Mixer Columns are selected before using **Save Mix As...**, a pop-up will appear, asking if you want to save just the selected columns. Press **Yes** to continue, or **No** to save the complete mixer.



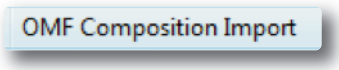
# XPro Import/Export

The **XPro Import/Export Functions** option allow audio data conversion to and from a variety of formats. At the moment the list of available formats includes: Fraunhofer MPEG Layer-3, Pro Tools Project, SDR Projects.



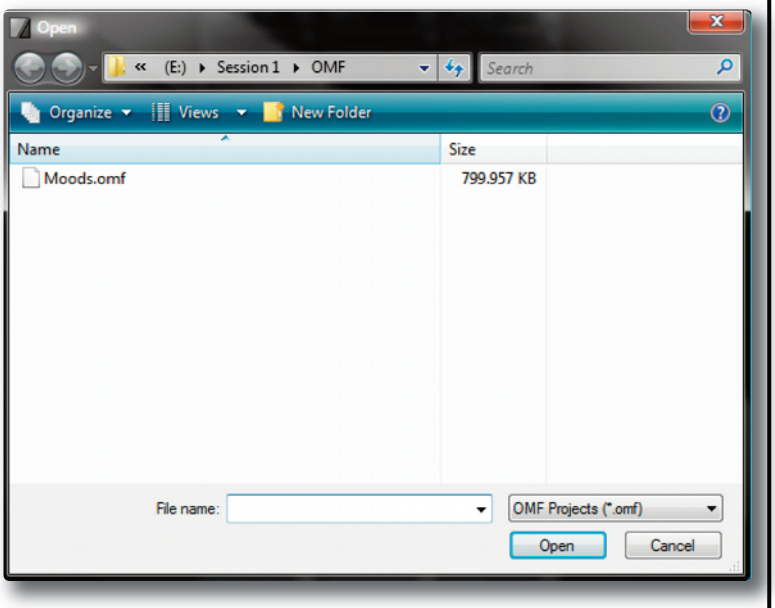

*Note that some elements of this menu are only present for legacy purposes. Import and export of a wider range of DAW formats is available via the **File** menu's **Save Arr and Pro-Convert** options. For more information, please refer to the 'Project Conversion' section.*

OMF Composition Import/Export and OMF Audio Import options are also available, for which separate software must be installed. All available XPro component options are organised in a submenu.

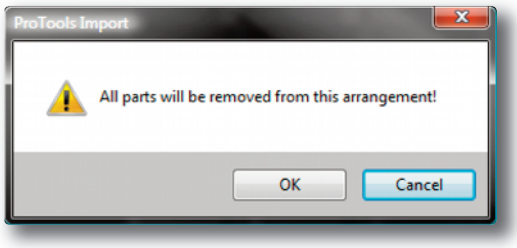


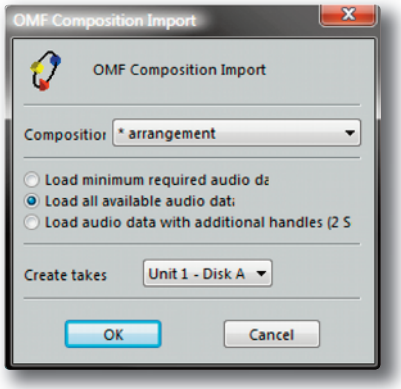

**OMF Composition Import**, **OMF Composition Export,** and **OMF Audio Import** options are available if you have installed the optional OMF Import/Export Plug-In Module. A password must be entered under the **Options** menu.

After selecting the **OMF Composition Import**, a standard Windows file browse dialogue appears, which you can use to locate and select an OMF Composition stored on any logical PC drive.




If there are any Parts in the Arrange window, a message box will appear and warn you that these Parts will be removed from the Arrangement.




Click **Cancel** if you need to save the current Arrangement first, otherwise click **OK**.

After selection, clicking **Open** will open a dialogue box (shown left) with audio import options:

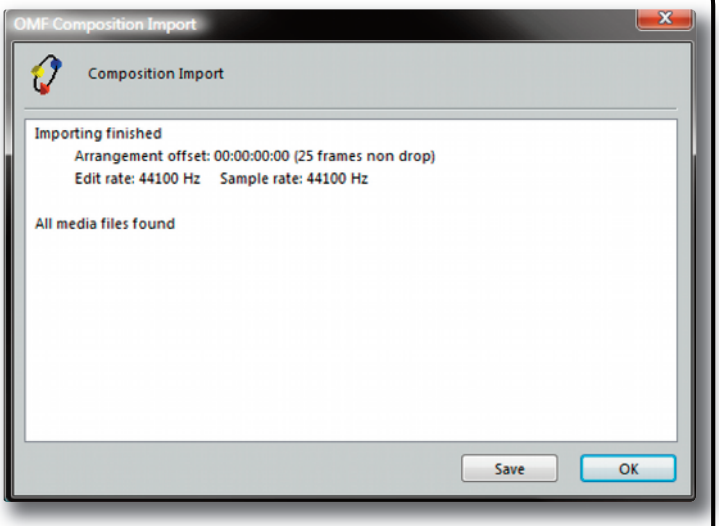
- **Load minimum required audio data** imports only the audio used by the Parts
- **Load all available audio data** imports the whole Takes even if the Parts are shorter
- **Load audio data with additional handles** imports the audio used by the Parts plus handles.

Clicking **OK** will start the import and a progress Window will be displayed.

After completion of the procedure a log Window will appear and present the results of the conversion.

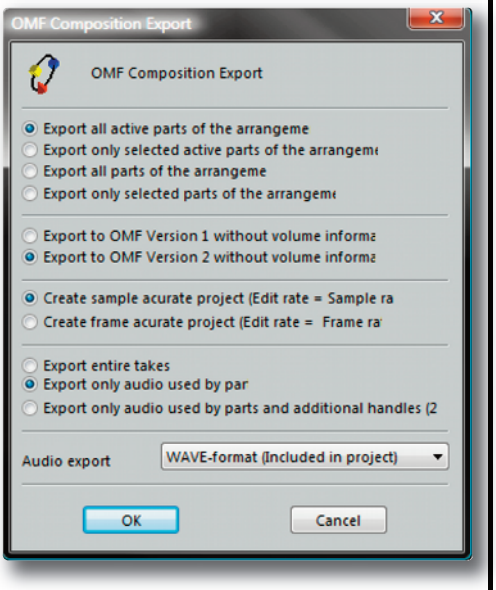
You can **Save** this log or click OK to finish the Import.

The audio files from the OMF Composition are loaded automatically as Parts at the right time positions inside the current Soundscape Arrangement, which will need to be saved in the usual way.




OMF Composition Export

Click on **OMF Composition Export** to open the OMF Export options.




**Top Section** Use the options to specify which Parts of the Arrangement should be exported, depending on their active/inactive and/or selected/unselected status. If you wish to select certain Parts for export, this should be done before you start the procedure.

**Second Section** Choose the OMF version of the OMF Composition which will be created.

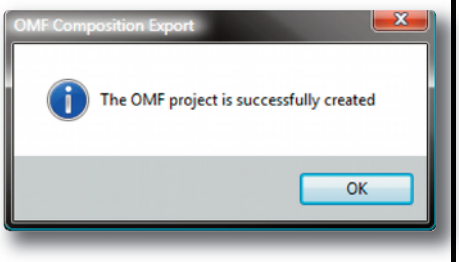
**Third Section** Specify the Edit Rate of OMF Composition, Sample or Frame Accurate.

**Fourth Section** Specify if a complete Take referenced by each exported Part or only the audio actually used by the Parts should be exported. The third option exports the used audio with additional 2 sec handles (before and after the used portion).

**Bottom section** Select the audio file format for the exported audio Takes. The available formats are WAV, AIFF and SDII. WAV and AIFF files can be stored separately or encapsulated inside the OMF composition. SDII files can only be stored separately.




Click **OK** to open a standard Windows File Browser and select the location to export to. Clicking **Save** will initiate the export and a progress Window will be displayed, shown left, followed by a Success Message when the Export is finished, shown right.

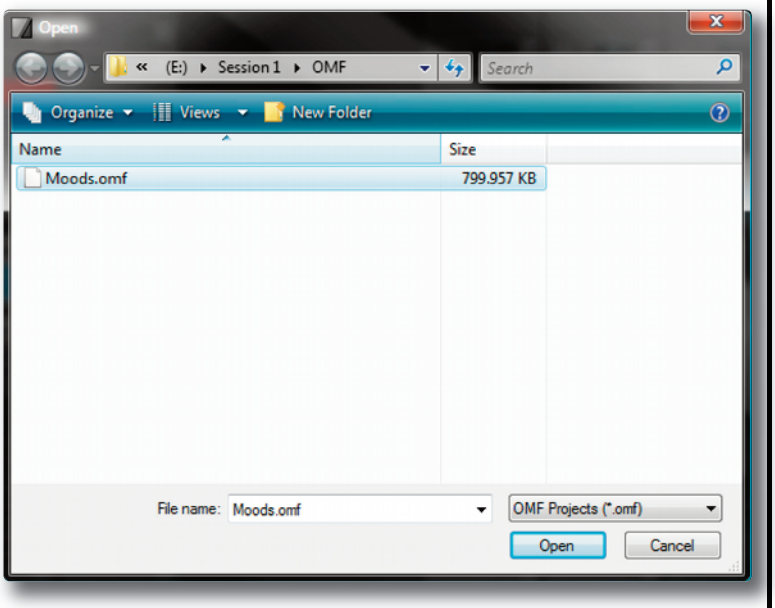



*NOTE: The Export tool cannot handle Arrangements which contain Part overlaps or crossfades, therefore it might be necessary to use the Flatten Arrangement **Ctrl+Shift+K** command in the Global Menu.*

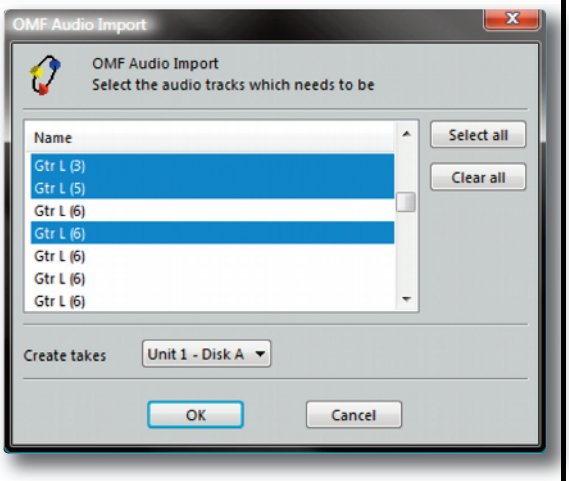
OMF Audio Import

The **OMF Audio Import** option lets you import the individual audio files from an OMF composition as Takes. These files are not loaded as Parts in the Arrangement but created as new Takes in the Recording/Process Folder.

Locate and select the OMF Composition from which you want to import audio. Even if the audio files are stored separately from the Composition, there is no need to select them. Select the OMF composition in all cases.




Click **Open** and select or deselect the files by clicking them individually, with or without using the ***Ctrl*** or ***Alt*** keys in the usual Windows way, or use the buttons to the right of the window to select or deselect all files in one operation.



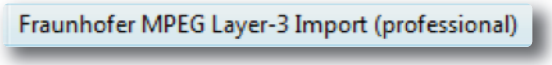

Click **OK** to start the import and a progress Window will be displayed.



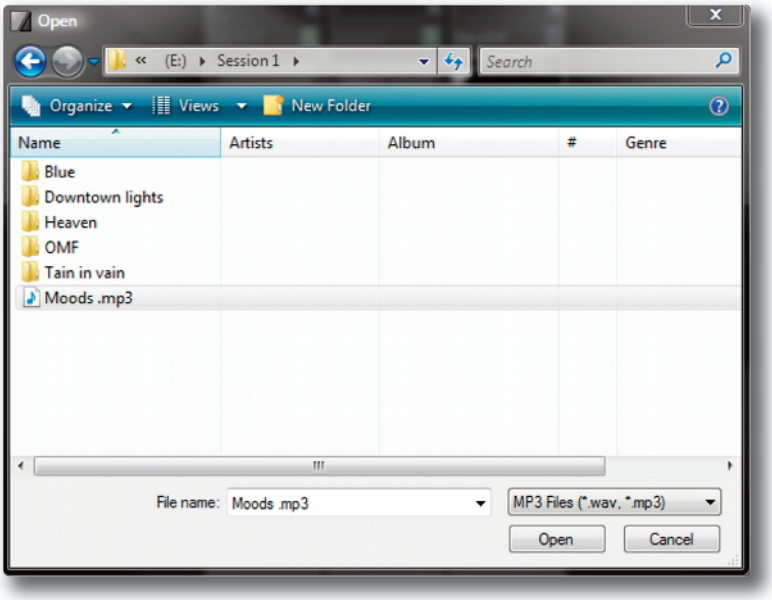

The imported files will be created as Takes on the active Record/Process folder.

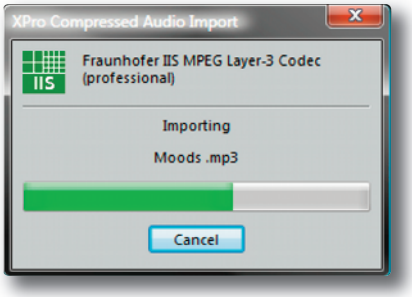
## MPEG Layer-3 Import and Export

*IMPORTANT: The Fraunhofer MPEG Layer-3 Import and Export option are only available if you have installed at least Windows Media Player 10.*
*If you only see the Import Option and it is without the (professional) extension, please first install a new Version of Windows Media Player (V10 or above) and then re-Install your Soundscape software (nothing will be lost!) to activate the MP3 Codec in Soundscape.*




To import an MP3 file as a Soundscape audio Take, select **Fraunhofer MPEG Layer-3 Import (professional)**, then locate and select any MP3 file stored on any PC drive. Several MP3 files can be selected for import in one single operation. Stereo MP3 files will be imported as a stereo pair of Takes with the same name and an L or R extension.




Click **Open**, and a Progress Indicator appears.

When the Import is done, new Takes have been created in your current Record/Process Folder.

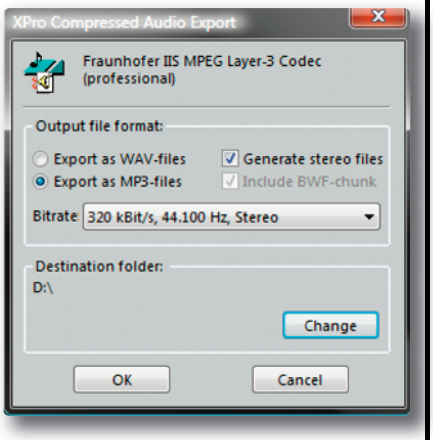
Fraunhofer MPEG Layer-3 Export (professional)

Selected Takes (all audio Takes selected in the File Manager) or Active Takes (all audio Takes referenced by Parts in the current Arrangement) can be enabled for export by selecting **Fraunhofer MPEG Layer-3 Export (professional)**, and clicking **OK** in the window which appears (shown right)



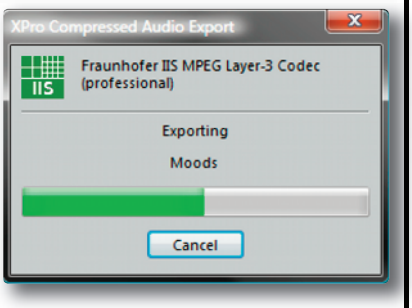

In the next Dialogue, the **Output file format** needs to be selected:

- **Export as WAV Files**: Allows to save as compressed MP3 Audio inside a .Wav Container.
- **Export as MP3 Files**: Saves the file inside a standard MP3 Container.
- Bitrate Selection: Select Bitrate, Sample Rate and Mono/Stereo Format options.
- **Generate Stereo Files**: Takes with same name and L/R extension will be exported as a Stereo MP3 File.
- **Include BWF Chunk**: A BWF-chunk is derived from the Take properties comments field and includes the original Time Stamp (Start Point). (Only available if **Export as WAV Files** is selected.)




In the bottom section you can view and Change the currently selected Destination Folder.

Click **OK**. to start the Export. A Progress indicator will be displayed until the Export is done.



## ProTools Session Import and Export

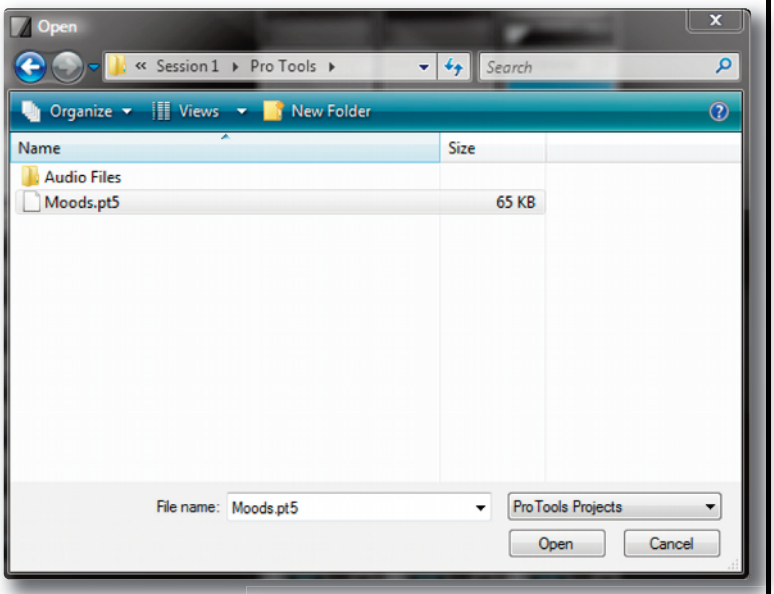
ProTools Project Import

The **ProTools Project Import** Function enables Soundscape to read AVID Pro Tools© Projects, created with earlier Versions of this DAW Software (V3.-V5.x).

*IMPORTANT: If you need to exchange Pro Tools Projects in a later Format (6,7,8.x) you can purchase SSL Pro-Convert, which translates Soundscape Arrangements into the latest Pro Tools Session formats and vice versa. For more Information about SSL Pro-Convert, please visit the product website at:*
*www.solidstatelogic.com/music/pro-convert/index.asp*
*Note that Pro-Convert is already embedded in Soundscape 6.2 and simply needs a password to unlock it.*

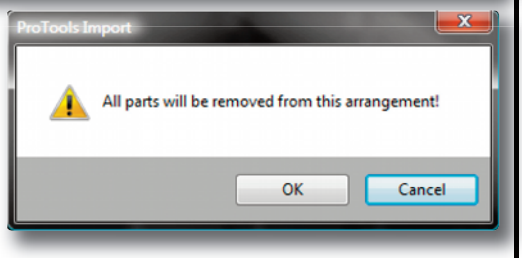
*NOTE: There is no need for a password to authorise the Pro Tools Import/Export function, even though there is a line for entering one in the Optional Modules dialogue box .*

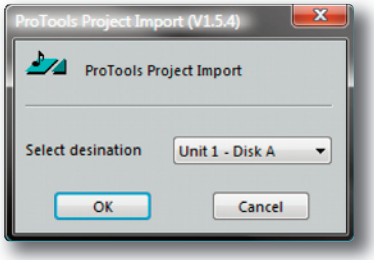
Click on **ProTools Project Import** and browse to the Session you want to import.




Select the Project and click **OK**.

If there are any Parts in the Arrange window, a message box will warn you that these Parts will be removed from the Arrangement. Click **Cancel** if you need to save the current Arrangement before proceeding, otherwise click **OK**.

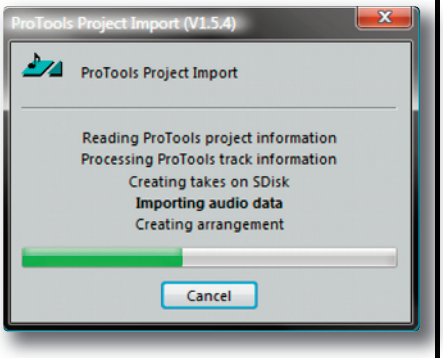



In Soundscape you don't have to select a Destination, the Import will go into the currently selected Record/Process Folder.

Click **OK**.

A Progress Window will appear until the Import is complete.




The audio files from the Pro Tools Project will be loaded automatically as Parts at the required time positions in the current Arrangement which will then need to be saved in the usual way.

## Additional information regarding Pro Tools Project Import

**Supported Pro Tools formats:**

- PT Sessions created on Mac OS: V3.x, V4.x, V5.0, and V5.1
- PT Sessions created on Windows: V5.0 and V5.1

**Preparing Sessions in ProTools for Import**

Click **Save Session Copy In** (**File** Menu) to create a copy of the Project in a new folder.

In the dialogue box, select the **Pro Tools 5.1 Session** format, **WAVE** audio file type, check the **Enforce Mac/PC Compatibility** option, and select the **All Audio Files** option in **Items to copy**.

The newly created project can be read from NTFS, FAT32, HFS-disk or Hybrid CD (HSF+/ISO9660) (HFS/HFS+ might require Software such as MacDrive or Mac Opener being installed on the PC).

All audio file formats are supported (WAVE, AIFF, SDII).

**Disk and File formats**

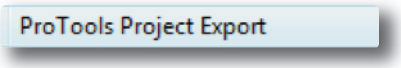
Some file information that Windows must be able to access resides in the Mac 'resource fork', especially when the Projects were created on legacy Mac OS Systems (OS 9).

- In case the files are stored on an HFS(+) volume, conversion to MacBinary format is required for SDII and V3.x, V4.x and V5.0 Mac-version project files to get single files that contain both data and resource fork.
- If the projects are stored on a FAT32 volume written by a Mac operating system, no conversion is required. The tool is able to find the files that contain the resource fork data.
- The 'MacDrive' application from MediaFour can read HFS(+) volumes in the Windows OS, and can convert files to MacBinary in order to make the resource fork readable by the Import tool.

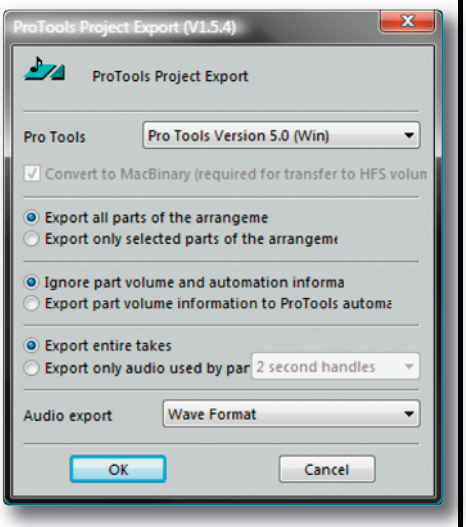
**Known limitations**

Pro Tools automation data cannot be imported at present.

*IMPORTANT: If you need to exchange Pro Tools Projects in a later Format (6,7,8.x) and also Import Volume/Pan Automation, Markers and Crossfades, you can purchase SSL Pro-Convert, which translates Soundscape Arrangements into the latest Pro Tools Session formats and vice versa. For more Information about SSL Pro-Convert, please visit the Product Website at: www.solidstatelogic.com/music/pro-convert/index.asp*




**ProTools Project Export** saves the current Arrangement as a legacy Pro Tools Session.




First, Specify the version of the target Pro Tools system (3.x, 4.x and 5.0 for Mac and version 5.0 for PC). If you select a Mac version, the **Convert to MacBinary** check box will be enabled. It must be checked if you are transferring to HFS volumes.

In the second section, you can decide to export either all the Parts in the Arrangement or only the selected Parts.

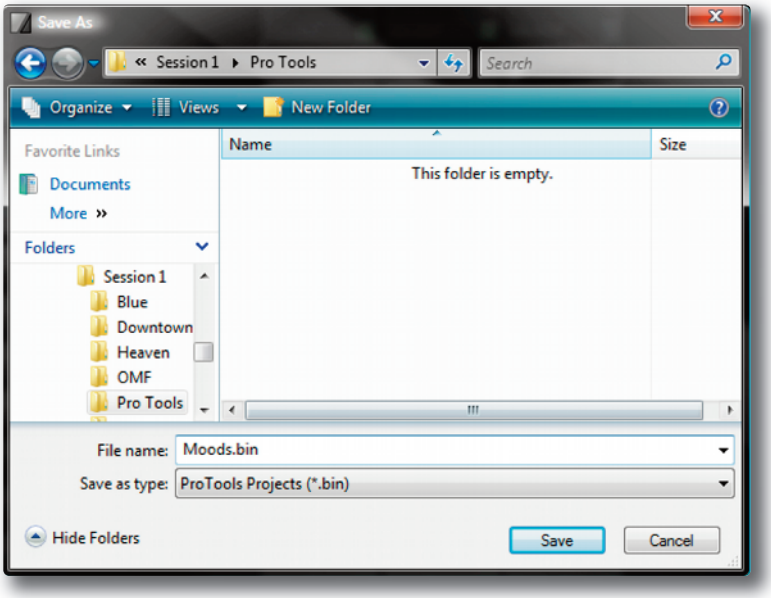
The third section allows you to ignore or translate Part Volume to Pro Tools automation data. Note that this is not available for Pro Tools V3.0 Mac.

In the fourth section, you can decide to export the complete Take referenced by each Part in the Arrangement, or only the audio actually used by the Parts, without handles or with 2, 5, 10 or 30 seconds handles. Handles can only be created if there is enough audio in the Take at each end of the Part.

In the fifth section, select **WAV** or **SDII** format for the exported audio.

Click **OK**.

Choose the Destination in the Windows File Browser and give the Session a name...




Click **Save**.

The Export will start and a Progress Window appears until the export is finished.



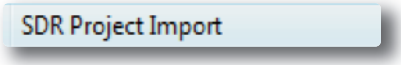

## Additional information regarding Pro Tools Export

### Known limitations

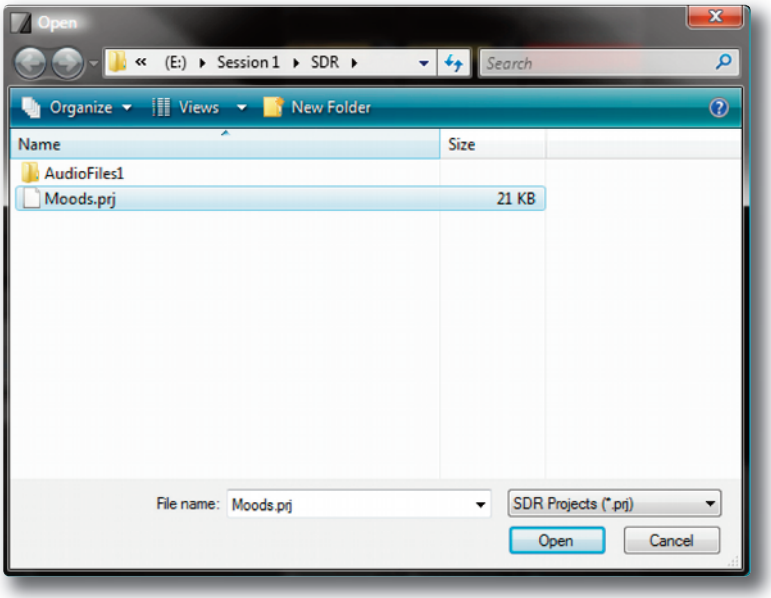
- Soundscape automation data cannot be exported to Pro Tools at present. However Part volume information is translated as Pro Tools automation data.
- The Export tool cannot handle Arrangements which contain Part overlaps or crossfades, you may need to Flatten the Arrangement and/or use the Consolidate Tool on crossfaded Part Sections before export.

*IMPORTANT: If you need to exchange Pro Tools Projects in a later Format (6, 7, 8.x) and also Export Volume/Pan Automation, Markers and Crossfades, you can purchase SSL Pro-Convert, which translates Soundscape Arrangements into the latest Pro Tools Session formats and vice versa. For more Information about SSL Pro-Convert, please visit the Product Website at: www.solidstatelogic.com/music/pro-convert/index.asp*

## SDR Project Import and Export

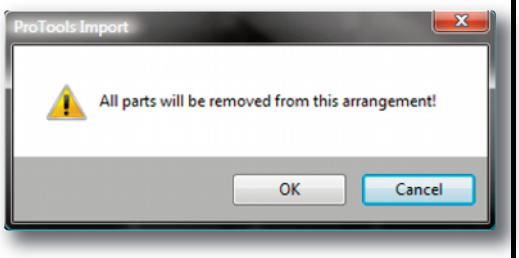


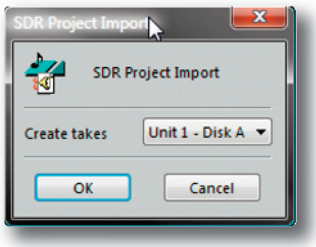

The **SDR Project Import** option can be used to locate, select and import an SDR Project (Mackie SDR/MDR/HDR24/96 format) stored on any PC logical drive.




Click **Open**.

If there are any Parts in the Arrange window, a message box will warn you that these Parts will be removed from the Arrangement. Click **Cancel** if you need to save the current Arrangement before proceeding, otherwise Click **OK**.

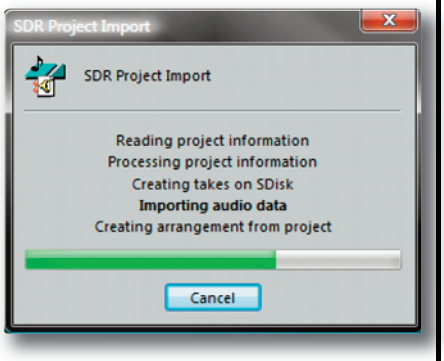


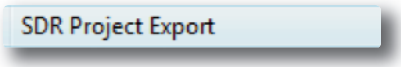

In Soundscape you don't have to select a Destination, the Project Import will go into the currently selected Record/Process Folder.

Click **OK**.

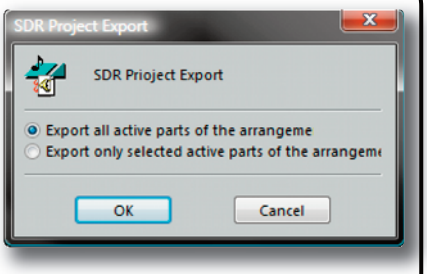
A Progress window will appear until the Import is done.



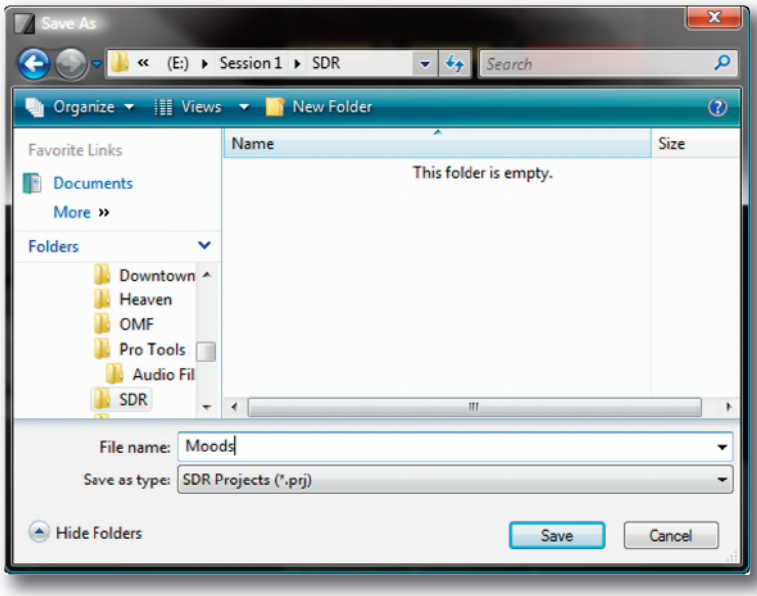

The audio files from the SDR Project will be loaded automatically as Parts at the right time positions in the current Soundscape Arrangement, which will then need to be saved in the usual way.




With the **SDR Project Export** option the current Arrangement or only the selected audio Parts can be exported to an SDR project (Mackie SDR/MDR/HDR24/96 format).




Click **OK**, choose the Destination in the Windows File Browser, give the Session a name, and click **Save**.




The Export will start and a Progress Window appears until the export is finished.




## Video Files




**Open Video File** allows to you select a video file stored on the PC for playback in Soundscape's Video File Player. By default the file will automatically play from the start of the current Arrangement, however an offset can be entered (**Settings** / **Video File Player** / **Setup...**). Other parameters for video playback can also be defined in this menu.

*Please refer to Chapter 4, 'Video File Player Window' for a detailed explanation.*




Select **Close Video File** to stop the Video File Player Module inside Soundscape. Closing/Hiding the Video File Player Window leaves the Video Playback active but invisible.

## Tempo Maps



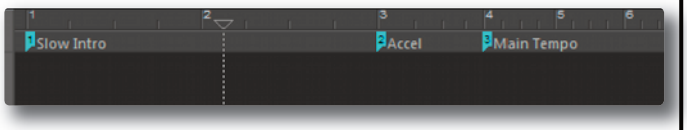

With **Open Tempo map** you can locate and select a standard MIDI file stored on the PC to import its tempo map information into the Arrangement.

A Tempo Map can contain tempo and time signature changes at any Song Position.

During playback, the tempo and time signature will change as necessary to follow the tempo map.

When a Tempo Map is loaded into Soundscape, both the Tempo and Sign settings are not selectable anymore (greyed out).

The Time Axis will reflect the changes when it is set to display musical time divisions (Bars/Counts/Ticks) and the Snap function will also take the changes into account.




Tempo Maps are especially useful when working in Sync with a MIDI Sequencer that is currently using Tempo and Signature changes .

Also, if you are working with Audio files that have been created with a Midi Sequencer (ie. Track Bounces of Virtual Instrument Tracks) using dynamic Tempo changes, the imported Tempo Map allows you to still edit with Snap in a musical grid.

In order to prepare a Tempo Map in your favourite Midi Application, please first consult its User Guide on how to create a standard MIDI File.

*IMPORTANT: Some sequencers provide a tempo map that has a slightly lower resolution than their internal tempo map. In such cases the midi file used to load the tempo map into Soundscape should be imported back into the sequencer as well.*




Clicking **Close Tempo map** will close the tempo map and Soundscape will return to a static Tempo and Sign, defined by the Tempo and Sign settings.

## Search File...

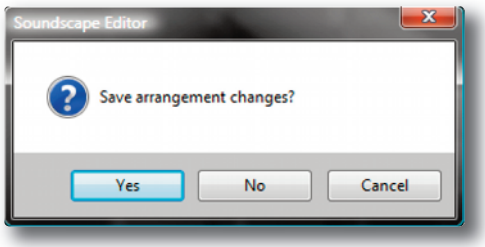


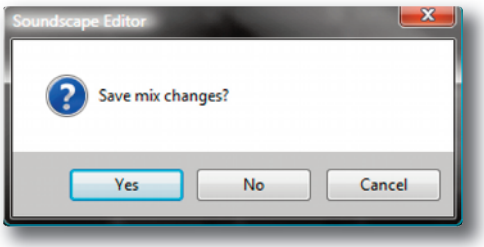

Selecting **Search File...** opens a pop-up in which you can define a search for a file. This function can also be accessed using the **Search** button in the top right-hand corner of the File Manager. Its function is described in full in the *File Manager* section of this guide.

## Exit



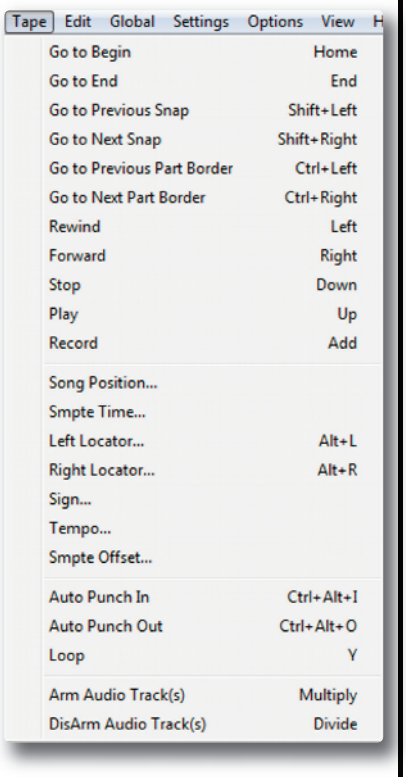

Clicking **Exit** will initiate the SSL Soundscape closing sequence. If any changes have been made to the active Arrangement and/or Mixer since the last time they were saved, either or both of the following dialogue boxes will appear, giving you options to save the current version of the Arrangement and/or mixer, not to save, or to cancel (ie., not to close the program):



# Tape Menu

The **Tape** menu is divided into four sections:




The items in the top section duplicate the function of the corresponding transport bar buttons.

Clicking on any of the items in the second section of the menu will open a dialogue box where a value can be entered for the corresponding parameter.

The **Auto Punch In**, **Auto Punch Out** and **Loop** modes can be enabled or disabled by clicking the items in the third section of the menu.

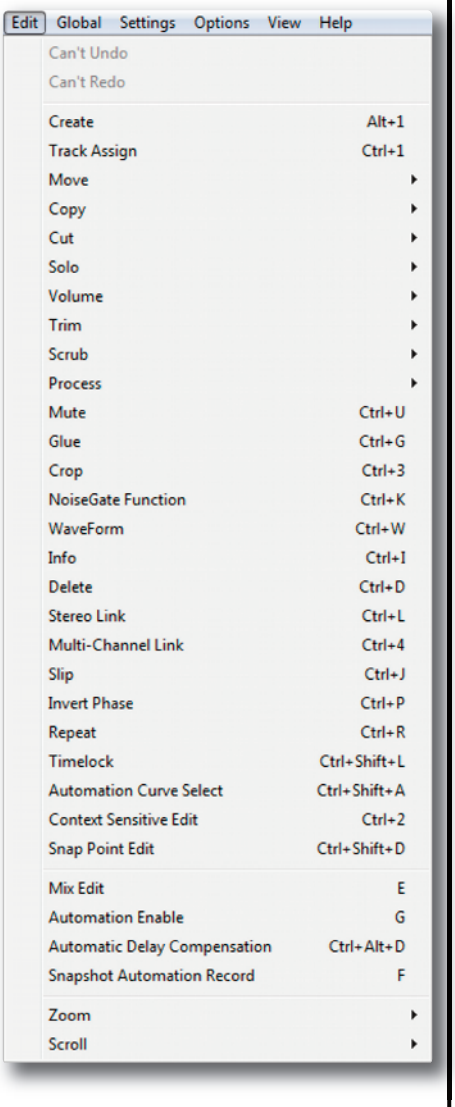
The items in the lower section of the menu can be used to arm or disarm all the existing audio record tracks.

*All items are explained in detail in Chapter 4: 'Arrangement Window.'*

*IMPORTANT: Certain items in this menu are unavailable in Play mode or Record mode. When this is the case they appear dimmed.*

# Edit Menu

The **Edit** menu is divided into four sections. The top section can be used to access the Undo/Redo function, the second section can be used to assign the editing tools to the mouse (left Mouse only, without ***Alt*** key) , the third section duplicates the toggle buttons on the bottom right of the Mixer Window (Edit Mode, Automation Enable, Automatic Delay Compensation and Snapshot), and the fourth section is dedicated to the 'Zoom' and 'Scroll' functions.




## Undo/Redo

Any edit that has been made to Parts or Takes can be undone or redone using this function. Up to 99 levels of Undo/Redo are available depending on the value specified in the **Settings** / **Preferences** / **General** / **Undo/Redo levels** menu.

**Undo** and **Redo** can also be performed using the ***Ctrl+Z*** and ***Ctrl+Y*** key combinations respectively.

Saving an Arrangement or closing Soundscape will clear the Undo/Redo Database and finalize the Take deletes the user has been confirming during editing.

## Editing tools

The second section of the menu lists the editing tools, and clicking on any item in this section assigns the corresponding tool to the left mouse button (except for the Context Sensitive Edit tool, which uses both mouse buttons incl. the ***Alt*** modifier key). Some of the tools are grouped in submenus according to their function.

All Tools are explained in detail in Chapter 4: *'Editing Tools'*.

## Mixer, Automation and Automatic Delay Compensation

The items in this section of the menu duplicate the four buttons in the bottom right corner of the Mixer.

These buttons have been described in detail in Chapter 4: *'Mix Window'*.

## Zoom and Scroll

When the mouse pointer is on 'Zoom', a submenu with six items appears. Clicking on either of the first two items selects the Zoom tool. Clicking the same item a second time will scroll through the memorised 'view sequence', as indicated by the '(1x=rect, 2xprev)' and '(1x=rect, 2xnext)' mentions. The next four items can be clicked to zoom in or out horizontally or vertically.

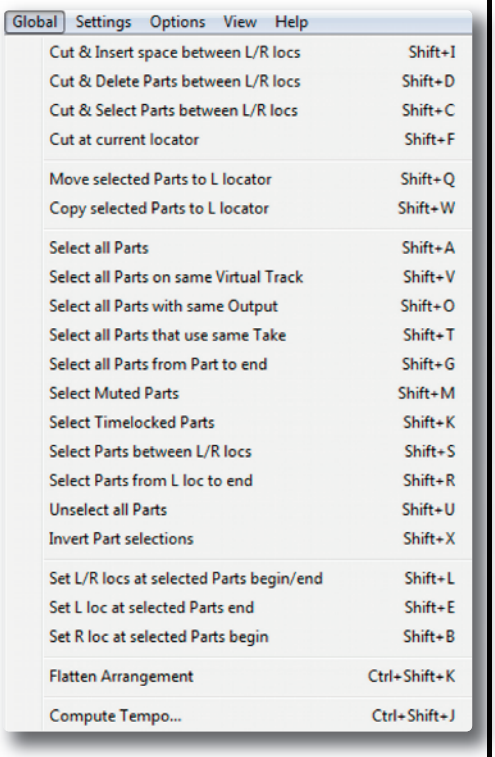
When the mouse pointer is on 'Scroll', a submenu with four items appears. These items can be clicked to scroll the Arrange window horizontally or vertically.

The operation of the Zoom tool, the zoom button, the 'view sequence/zoom history' and Scroll functions have been described in detail in Chapter 4: *'Arrange Window': 'Arrangement Navigation'*.

***IMPORTANT:*** *Certain items in this menu are unavailable in Play mode or Record mode. When this is the case they appear dimmed.*

# Global Menu

The **Global** menu provides access to some Global Commands or editing Macros. Most of these functions apply to the whole Arrangement, or to a group of selected Parts. Many of them are conditioned by, or affect the Left and Right Locator positions, and all can be accessed quickly using a combination of the ***Shift*** key plus a hot key (as well as from the menu itself).



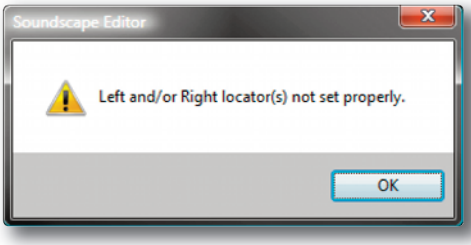

Since all Global functions can seriously accelerate the editing workflow, it is a good, if not very good idea, to not only learn how they work but also to memorise and use the keyboard shortcuts.

Many recurring editing tasks, requiring 3 or more steps, can be done using the Global Commands with a single keystroke!

For all the functions that are conditioned by the position of both Locators, like:

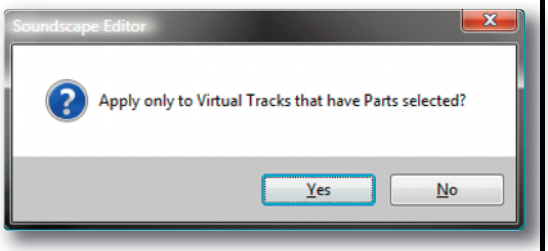
- Cut & Insert space between L/R locs
- Cut & Delete Parts between L/R locs
- Cut & Select Parts between L/R locs
- Select Parts between L/R locs

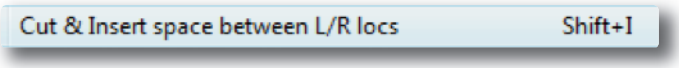
...if the Locators are set incorrectly (ie. Left Locator is placed before the Right Locator), when trying to use the function the warning shown right will be displayed:




Click **OK** and reposition the Locator(s) and perform the Global Command again.

For some of the global functions, if any Part or group of Parts is selected before the edit is made, a dialogue box will be displayed, which lets you decide if the edit should be restricted to the virtual track(s) that Part or group of Parts is on.

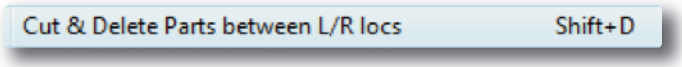



With **Cut & Insert Space between L/R locs**, all the Parts (or selected Parts) that lie across the Left Locator are cut at that Locator's position. All the Parts to the right of the Left Locator are then moved to the Right Locator.



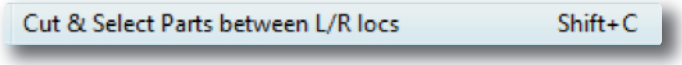

This function can be restricted to certain virtual tracks by selecting at least one Part on each targeted virtual track before performing the edit. If the Shift Markers with Global Insert/Delete option is active (**Settings** / **Preferences** / **Arrangement**) any Markers after the Left Locator will be moved with the Parts.




When **Cut & Delete Parts between L/R locs** is used, a cut is made to all Parts (or selected Parts) at the Left and Right Locator positions. The Parts between the Locators are then deleted, and all the Parts to the right of the Right Locator are moved as a group to the Left Locator, closing up the space created by the edit.



This function can be restricted to certain virtual tracks by selecting at least one Part on each targeted virtual track before performing the edit. If the Shift Markers with Global Insert/Delete option is active (**Settings** / **Preferences** / **Arrangement**) any Markers after the Left Locator will be moved with the Parts.




**Cut & Select Parts between L/R locs** creates a cut to all Parts (or selected Parts) at the Left and Right

Locator positions. The Parts between the Locators are then selected (ie. to Move/Copy them to a different Track)



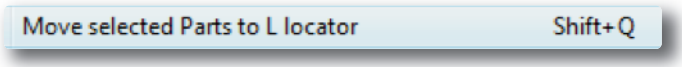
This function can be restricted to certain virtual tracks by selecting at least one Part on each targeted virtual track before performing the edit.




When **Cut at current locator** is used, a cut is made to all Parts (or selected Parts) at the Current Locator position.



This function can be restricted to certain virtual tracks by selecting at least one Part on each targeted virtual track before performing the edit.




When **Move selected Parts to L locator** is used, any selected Part(s) are moved to the Left Locator.



Copy selected Parts to L locator Shift+W

And when **Copy selected Parts to L locator** is used, any selected Part(s) are copied to the Left Locator:



Select all Parts Shift+A

**Select all Parts** will select any Part in the Arrangement.



Select all Parts on same Virtual Track Shift+V

In order to use the **Select all Parts on same Virtual Track** function, if a Part is selected, all Parts that are on that virtual track will be selected:



Select all Parts with same Output Shift+O

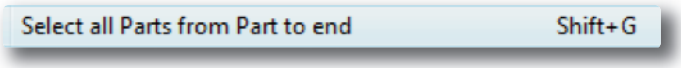
**Select all Parts with same Output** will select all Parts that have the same track assignment as the selected one(s). At least one Part needs to be selected first.



Select all Parts that use same Take Shift+T

In order to use the **Select all Parts that use same Take** function, at least one Part should be selected first. All the Parts that reference that Take will become selected.






In order to use the **Select all Parts from Part to end** function, at least one Part should be selected first. All the Parts that are positioned to its right, or that have the same beginning position, will be selected:

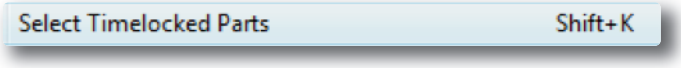


*NOTE: This function can also be used by holding **Ctrl+Shift** before selecting a Part.*




**Select Muted Parts** will indeed select all the muted Parts.






**Select Timelocked Parts** will select all timelocked Parts, as identified by the asterisk (*) appended to their name.

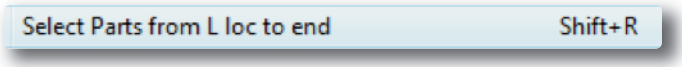


*NOTE: Please refer to 'Timelock tool' of the 'Editing Tools' section for a detailed explanation of timelocked Parts.*




**Select Parts between L/R locs** will select all the Parts of which any section is positioned between the Locators:






**Select Parts from L loc to end** will select all Parts of which any section is positioned between the Left Locator and the end.



This function can be restricted to certain virtual tracks by selecting at least one Part on each targeted virtual track first.

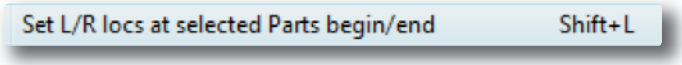



**Unselect all Parts** cancels all Part selections.



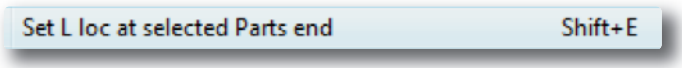

**Invert Part selections** deselects all selected Parts and unselected Parts are selected instead.





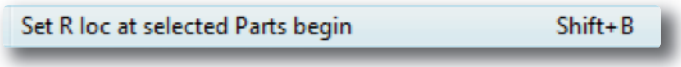

**Set L/R locs at selected Parts begin/end** positions the Left and Right Locators at the beginning and end respectively of the selected Part or group of Parts. This is especially useful when an area needs to be defined for using the Mixdown tool or a loop is selected to define the Tempo with Compute Tempo function (see next Page).






**Set L loc at selected Parts end** sets the Left Locator to the end of the selected Part (Part X2 in the example below) or group of Parts. This is especially useful when an auto punch in point needs to be set precisely set at the end of a Part.






**Set R loc at selected Parts begin** sets the Right Locator to the beginning of the selected Part (Part X3 in the example below) or group of Parts. This is especially useful when an auto punch out point needs to be set precisely at the beginning of a Part.





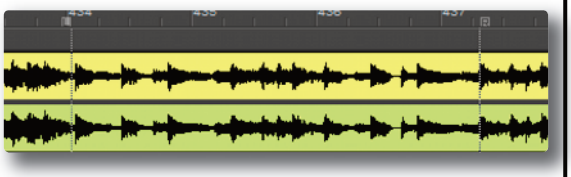


**Flatten Arrangement** will remove any overlapped Parts, or overlapped sections of Parts and crossfades from the Arrangement . If any Takes become unreferenced as a consequence however, they are not deleted. This is useful for exporting Arrangements to certain formats or with export modules where real time crossfades and overlapping parts are not supported.

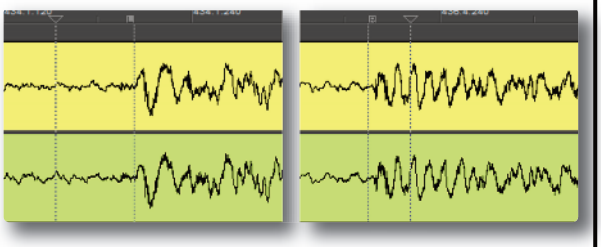



**Compute Tempo** calculates the tempo for Soundscape based on the Left and Right Locator positions, the current Time Signature and the number of bars being considered.

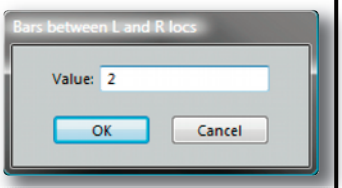
In the following example, it is used to compute the tempo of a drum Part.



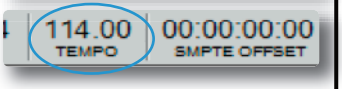
- While the recorded audio Part is played back, drop in the Left and Right Locators on the fly (by pressing the ***L*** and ***R*** key) to mark the beginning of bars.
  In our example, the Locators are dropped two bars apart, but any number of bars can be used.



- Stop Playback and zoom in to a sufficient level and clicking on the Left and Right Locator buttons in the Transport Bar (blow the readouts) it is easy to jump between the two marked points.
- It is now possible to place the Left and Right Locators with more precision by visual reference to the waveform. The two marked L/R locator Positions should look like this.
- **Sign** in the Transport Bar must match the time signature of the audio (ie. 4/4 for Four on the Floor).
- Click **Compute Tempo** and enter the number of bars between the Locators in the following dialogue box.

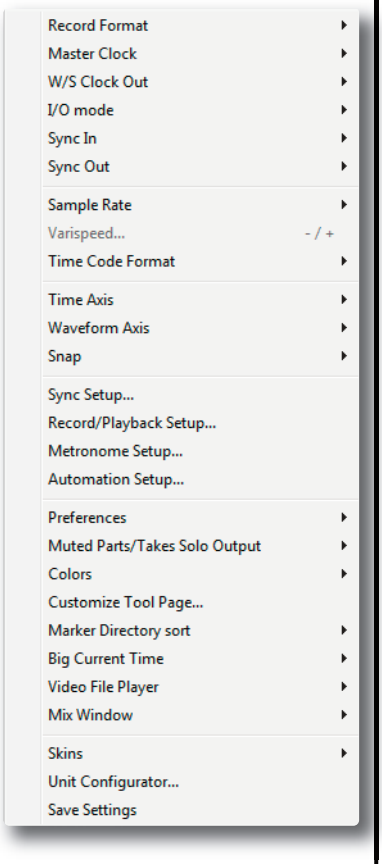



- Click **OK** and the tempo is instantly calculated by the software and the value in the **Tempo** readout is updated accordingly.



# Settings Menu

The **Settings** menu allows you to configure your Soundscape system in the way you prefer to work, adapt to how your studio environment is set up and configure global parameters as well as Arrangement Settings. It includes all kinds of settings related to the operation of the hardware and software, display preferences, etc.




Global Setting changes in this menu are temporary and will revert to your preferred defaults on next program launch.

The last item in the menu, **Save Settings**, however, saves all current global settings as default.

Arrangement Settings are only remembered when the Arrangement has been saved.

Since a number of the settings accessed from this menu have been explained in detail in other chapters, a reference will be made where appropriate (hyperlinked to the related page).

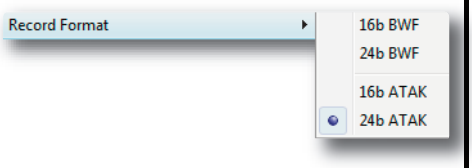
## Record Format

Soundscape can record audio as Broadcast Wave File (extended Version of the standard WAV Format) or as SSL Soundscape ATAK Audio File (SSL Soundscape's proprietary audio file format) in 16-bit or 24-bit at the projects Sample Resolution, for new recordings.

24-bit provides increased dynamic range compared to 16-bit, and therefore also more headroom. 16-bit however requires 33% less disk space.

The BWF format has been defined by the Audio Engineering Society (AES) and the European Broadcasting union (EBU).

BWF 'Broadcast Wave Format' files are WAV files with a Broadcast Extension chunk included. It is a good choice if you need compatibility for sharing files with other software packages.




The ATAK format offers an embedded unique ID for every recorded or processed audio file, therefore Soundscape will find the Unique audio files for an Arrangement regardless of their location (within the user definable range of Domain Root Folders).

This makes the ATAK format ideal for projects where a large number of files are handled, collaboration between different Soundscape Seats is mandatory or whenever you need to work on a dynamically changing storage system.

Also the ATAK Format is not readable by any other Audio Software, making it a safe foundation for larger facilities or certain departments.

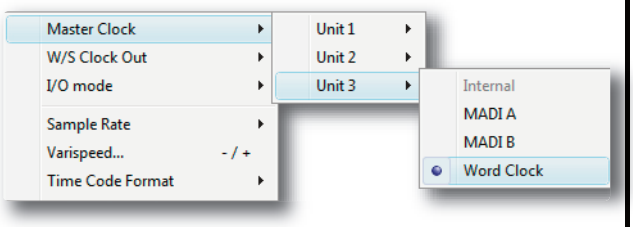
The recording format can also be set in the Status Bar.

*NOTE: The Recording format can be changed during a production at any time and only new Recordings and Mixdowns will be created in the newly selected format. This allows to quickly change ie. to 16-bit BWF, mix down a stereo master and use the resulting file to burn a CD.*
*Existing Takes referenced by Parts will NEVER be changed using this function!*

# Master Clock

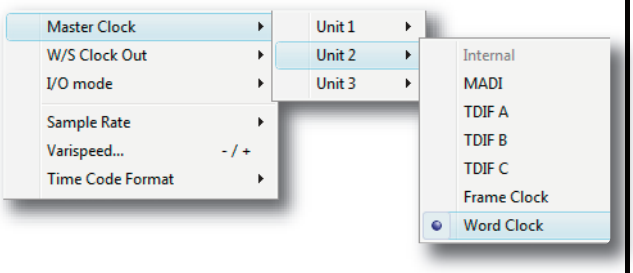
The Master Clock can be altered for each Soundscape Unit separately, that is part of the Unit Configuration. Available clock sources vary depending on device type (or Alpha-Link interface a Mixpander Card is connected to).

## Master Clock settings for an MX4




- **Internal**: The Master Clock is generated by MX4's on-board clock generator
- **MADI A or MADI B**: MX4 derives its Master clock from the MADI Input A or B, which carries an embedded Clock inside the MADI stream
- **Word Clock**: Generally suitable for slave synchronisation to a Word Clock signal from any other device. Especially in bigger facilities the Wordclock will be generated by a House Clock Generator, that has multiple direct Word Clock Outputs and syncs the whole facility to this stable clock in a star distribution.

### Master clock settings for a Mixpander and SSL Alpha-Link

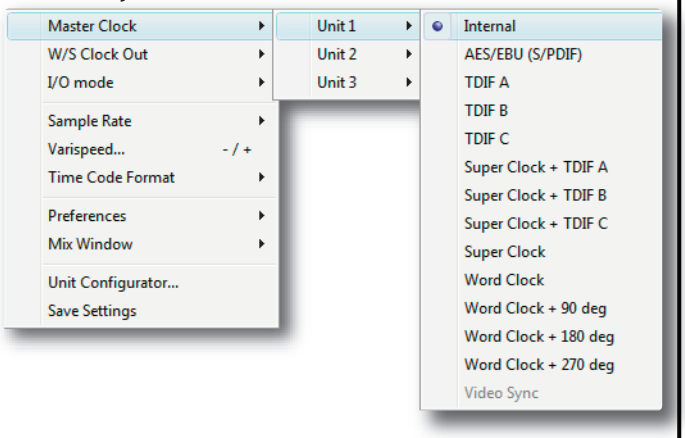



Since Mixpander systems can be connected to various models of the SSL Alpha-Link (or legacy Soundscape iBox Product Range) with several digital I/O options, the available Master Clock options vary (and only depend on the I/O configuration of the connected converter box).

- **Internal**: The Master Clock is provided by the on-board clock generator of the Alpha-Link or iBox device
- **Digital Input Clock Options**: This could be MADI, ADAT, AES/EBU or TDIF, depending on model and configuration.
- **Frame Clock**: Clock signal similar to Word Clock, used when the actual Sample Rate is a multiple of the transmitted audio frame size. Frame Clock is used at high Sample Rates for digital signal formats, that can not be clocked higher than standard Rates.
  For example one ADAT Frame can contain 8 Channels of 24-bit Audio at 48 kHz. At 96 kHz the ADAT Frame can still carry the same amount of data. To achieve double the amount of data per channel, a 96 kHz ADAT Frame only contains 4 Channels of 24-bit Audio (achieved through 'sample multiplexing', sometimes termed 'SMUX'). The Clock (or number of Frames/second) stays at 48000 ADAT Frames/s, therefore the Frame Clock remains 48 kHz even when running the System at 96 kHz.
- **Word Clock**: Generally suitable for slave synchronisation to a Word Clock signal from any other device. Especially in bigger facilities the Wordclock will be generated by a House Clock Generator, that has multiple direct Word Clock Outputs and syncs the whole facility to this on stable clock.

*NOTE: The Master Clock on Mixpander Systems is always generated or received by the Alpha-Link or iBox interfaces. The card itself can neither generate a Master Clock signal itself nor read any of the industry standard external Clocks formats.*

## Master clock settings for a Mixpander connected to Soundscape 32 unit



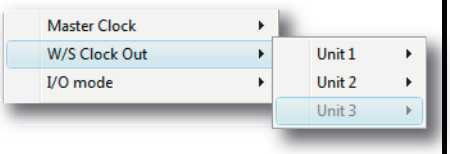

- **Internal**: When this option is selected, the Sample Rate is set by the Soundscape unit's on board Master Clock Generator. This should be used in conjunction with analogue equipment only.
- **AES/EBU (S/PDIF)**: When this option is selected the Soundscape 32 is slaved to the WordClock signal received at its AES/EBU (S/PDIF) input.
- **TDIF A, TDIF B, TDIF C**: When one of these options is selected Soundscape 32 operates as WordClock Slave to the source device connected to the corresponding TDIF port.
- **Super Clock + TDIF A, Super Clock + TDIF B, Super Clock + TDIF C**: These options operate in a similar manner to the TDIF A, TDIF B, and TDIF C options described above, except that a SuperClock signal at 256 times the Sample Rate is used to provide the higher frequency system clock for Soundscape 32, with the LRClk signal providing the WordClock reference.
- **Super Clock**: If this setting is selected, it is assumed that Soundscape 32's internal clock divider produces the sample frequency Word Clock derived from the incoming Super Clock and that other devices are slaved to Soundscape 32 via the Word Clock, TDIF, or AES/EBU (S/PDIF) outputs. Super Clock is a higher frequency clock signal (256 x Sample Rate).
- **Word Clock**: When this option is selected Soundscape 32 is slaved to the WordClock signal received at its WordClock input connector.
- **Word Clock + 90/180/270 deg phase**: These variants of the 'WordClock' setting are provided to adapt to different TDIF implementations as found in other devices, such as, for example, the Mackie D8B digital mixing console.
- **Video Sync**: This option is only available when the optional Soundscape Synchronization Board is installed in the Soundscape 32 unit (Unit 1 in a multiple unit system). If it is selected Soundscape 32 can be slaved to any Video Synchronisation signal such as Blackburst or Composite received via the VITC Input of Synchronisation Board.

## W/S Clock Out

With this menu entry you can control the Clock signal to be transmitted at the Word Clock / Super Clock output connector of the installed device(s). As mentioned above, the available options depend on the installed SSL components.

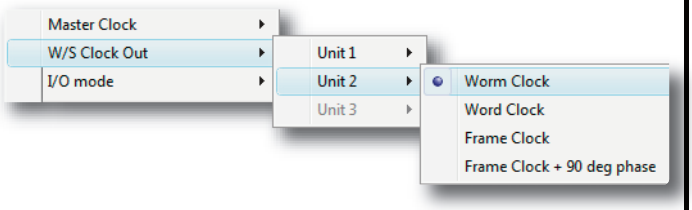
### W/S Clock Out settings for MX4

The MX4 has one BNC Word Clock connector, which can transmit or receive Word Clock Signals. The Word Clock Connector can only Output Word Clock signals, when the Master Clock is set to Internal. If the Master Clock is set to MADI or Wordclock, the W/S Clock Output selection is greyed out.



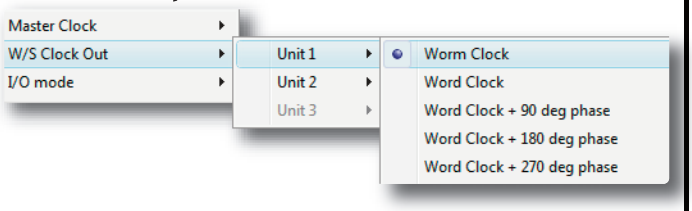

### W/S Clock Out settings for a Mixpander and Alpha-Link/iBox units

For Mixpander cards, the W/S Clock Out signal is actually transmitted by the connected iBox 24/48/64 or SSL XLogic Alpha-Link interface.




- **Worm Clock**: SSL's proprietary extension of the Word Clock Format. Worm Clock is a modified Word Clock signal that includes sample accurate start/stop synchronisation information for several units. The Master Clock parameter for the slave unit must be set to Word Clock, which also enables Worm Clock slave synchronisation.
- **Word Clock**: When this option is selected the Mixpander System is slaved to the Word Clock signal received in the connected Alpha Link or iBox Interface.
- **Frame Clock**: Clock signal similar to Word Clock, used at high Sample Rates for digital signal formats, that can not be clocked higher than standard Rates. (see above for more details)
- **FrameClock + 90 deg phase**: Must be selected when connecting certain devices whose Audio signals and clock signals are out of phase. (mainly MADI or ADAT SMUX signals on devices from certain vendors)

### W/S Clock Out settings for a Mixpander connected to a Soundscape 32 unit



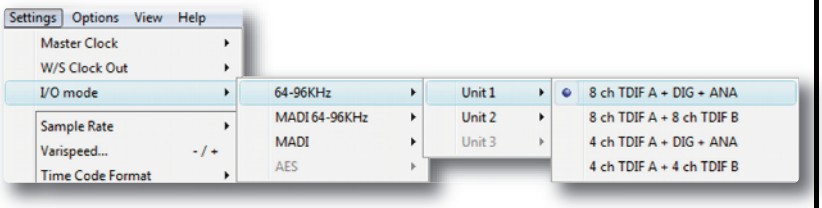

- **Worm Clock**: SSL's proprietary extension of the Word Clock Format. Worm Clock is a modified Word Clock signal that includes sample accurate start/stop synchronisation information for several units. The Master Clock parameter for the slave unit must be set to Word Clock, which also enables Worm Clock slave synchronisation.
- **Word Clock**: When this option is selected the Mixpander System is slaved to the Word Clock signal received in the connected Soundscape 32 System.
- **Word Clock + Phase**: When this option is selected the Mixpander System is slaved to a Word Clock signal received at the Soundscape 32 System, that is shifted against the Audio signal by 90°, 180° or 270°. Especially older digital devices like early DTRS Tape Recorders and some digital Consoles need those settings.
- **Super Clock**: When you have problems (such as digital clicks) providing only Word Clock to a slave device or of that Word Clock is varying in frequency (in Chase Slaving conditions), please use Super Clock instead.

# I/O mode

I/O Mode covers all hardware specific settings that define the I/O format configuration. This includes I/O Modes for operation at high Sample Rates and the general Settings for MADI Communication. Again, the available options are determined by the SSL Hardware you are using.

## 64-96kHz

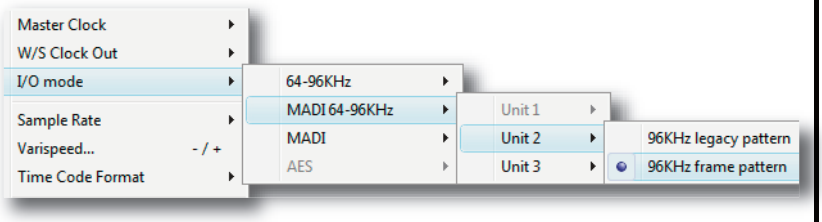
Allows selection of different I/O configurations and interleaving options at high sample rates, such as SMUX ('Legacy') or non-SMUX ('High Speed') Modes for ADAT/TDIF/MADI/AES or different I/O combinations for a Soundscape 32/Mixpander system.




## MADI 64-96kHz

In this submenu the MADI Frame Pattern is selected between 96k Legacy Pattern (48k Frame) and 96k Frame Pattern.

The options determine how the 28/32 MADI Channels are encoded inside one MADI Frame.

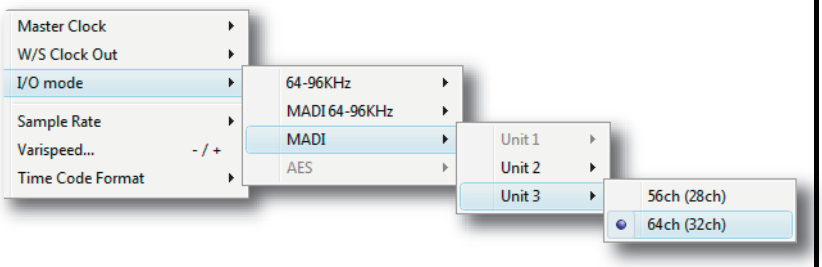



## MADI

This submenu selects the MADI CH Mode (56 or 64 Channels). In it's original inception MADI provided 56 Channels (at 44.1 kHz and 48 kHz) and some data band-width headroom to enable 'Vari-Speed' on all Channels at a higher play speed (=higher Sample Rate).

Since providing the facility for Vari-Speed is not necessary in many environments, a 64 Channel MADI Mode was introduced as an AES standard, which uses the full MADI data bandwidth, hence doesn't allow a higher sample rate (or Vari-Speed with positive Values).

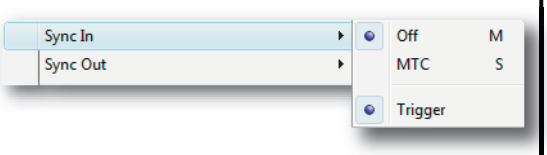
Both standards are equally popular, depending on the variety of applications and industries you work in, you may need to switch between these modes quite often. At 88.2 kHz and 96 kHz a 56 Channel MADI cable operates with 28 Channels of audio, while with 64 Channel MADI, 32 Channels of audio are transmitted.



## Synchronisation

Sync In defines whether Soundscape responds to external Midi Time Code, and Sync Out allows MTC to be sent from Soundscape. Sync configuration is performed in the **Settings** / **Sync Setup** window.

### Sync In




- **Off** (***M***): Clicking this option or pressing ***M*** will select it. When **Off** is selected, Soundscape does not respond to incoming Time Code.
- **MTC** (***S***): Clicking this option or pressing ***S*** will select it. In **MTC** Slave mode Soundscape 32 responds to MIDI Time Code received at its MIDI In port.

*NOTE: Soundscape will only respond to incoming MTC if it is in Play mode or Record mode, ie., if the Play or Record button has been clicked, or if the **Auto Play/Stop when slave syncing** option is selected under **Settings** / **Sync Setup**.*

### Sync Out

- **MTC**: This setting should be selected (by clicking on it or pressing the ***M*** key on the computer keyboard) to use SSL Soundscape as a MIDI Time Code Master, with one or more Time Code Slave devices synchronised to the MIDI Time Code sent via the MIDI Out port.

## Legacy Soundscape 32 Synchronisation

This section describes the synchronisation functionality of a legacy Soundscape 32 Unit connected to a PCI Host IF Card and configured as Unit1. Using a Soundscape 32 Unit bi-directional Sync to external devices as MTC Master, MIDI Clock Master, MTC Chase Slave or MTC Trigger Slave using its MIDI In and MIDI Out connectors is possible. Soundscape 32 can also lock to incoming LTC, VITC or 9-Pin Time Code, and output LTC and BITC, providing an integrated solution suitable for most synchronisation tasks in Music, Post and Broadcast. Since the Soundscape 32 Hardware can output several types of Time Code simultaneously while chase slaving to incoming Time Code, it can even act as a Time Code converter.

### Additional Sync In options

- **LTC** (***O***): Clicking this option or pressing ***O*** will select it. In **LTC** Slave mode Soundscape 32 respond to Longitudinal Time Code received via the LTC In port on the optional Soundscape Synchronisation Board (RCA/Cinch connector). If the Soundscape Synchronisation Board is not present the **LTC** option appears dimmed.

*NOTE: Soundscape will only respond to incoming LTC if it is in Play mode or Record mode, ie., if the Play or Record button has been clicked, or if the Auto Play/Stop when slave syncing option is selected under **Settings** / **Sync Setup**.*

- **VITC**: Clicking this option will select it. In **VITC** Slave mode Soundscape 32 respond to Vertical Interval Time Code received via the VITC In port on the optional Soundscape Synchronization Board (BNC connector). If the Soundscape Synchronization Board is not present the **VITC** option is not available and appears dimmed.

*NOTE: Soundscape will only respond to incoming VITC if it is in Play mode or Record mode, ie., if the Play or Record button has been clicked, or if the Auto Play/Stop when slave syncing option is selected under **Settings** / **Sync Setup**.*

- **9-Pin**: This option is available when SSL Soundscape is set to Controller, Synchro or Layback under menu: **Settings** / **9-Pin Operation** / **Mode**. Clicking it will select it. In **9-Pin** Time Code mode Soundscape 32 respond to 9-Pin Time Code received via the 9-Pin connector on the optional Soundscape Synchronization Board. If the SSL Soundscape Synchronization Board is not present the **9-Pin** option is not available and appears dimmed.

*NOTE: Soundscape will only respond to incoming 9-Pin Time Code if it is in Play mode or Record mode, ie., if the Play or Record button has been clicked, or if the Auto Play/Stop when slave syncing option is selected under **Settings** / **Sync Setup**.*

- **Chase**: This setting should be selected (by clicking on it or pressing the ***S*** key on the computer keyboard) to use SSL Soundscape as a Time Code Chase Slave. In this case, Soundscape will wait for incoming Time Code, and lock to that Time Code within about 1.5s (please also read the *Sync Setup* section of this chapter regarding the Sync In 'Preparation Time').

*NOTE: Soundscape will only respond to incoming Time Code if it is in Play mode or Record mode, ie., if the Play or Record button has been clicked, or if the Auto Play/Stop when slave syncing option is selected under* ***Settings*** */* ***Sync Setup****.*

Soundscape 32 implements a full '**Chase** Lock', which means that it continuously monitors the incoming Time Code, and if that time code presents fluctuations, Soundscape 32 will vary its sample rate to maintain optimum synchronization. This is essential if synchronizing to a tape machine that has Time Code recorded on one track, as the tape speed can vary and, without **Chase** Lock, significant timing errors would occur. Soundscape will also follow large changes in timing which occur when using varispeed, etc.

*IMPORTANT:*
*In Chase Slave mode, Soundscape's Sample Rate Master Clock is varied to adapt to any fluctuation of the incoming Time Code. Therefore it is only possible to record or play back via the digital I/O if the devices that are connected to the digital I/O are slaved to Soundscape 32's Master Clock.*
*If you are RECORDING audio such as LIVE musical performances while 'CHASING' a tape machine or other source that may produce large changes in Time Code, you are likely to hear pitch changes on playback when using a stable Time Code source (eg., MIDI sequencer) or SSL Soundscape's own internal clock.*

- **Trigger**: When Time Code Trigger Slave mode is selected, the received Time Code only indicates the start and stop locations and the Sample Rate used can either be internal, or received from a digital input as required. This means that when the Time Code Master is an external digital device which provides both a Time Code signal and a digital audio signal (such as a multitrack digital tape recorder that transmits MTC), full lock-up is provided by the digital input signal once playback or recording has been triggered via the Time Code.

*NOTE: Soundscape will only respond to incoming Time Code if it is in Play mode or Record mode, ie., if the Play or Record button has been clicked, or if the Auto Play/Stop when slave syncing option is selected under* ***Settings*** */* ***Sync Setup****.*

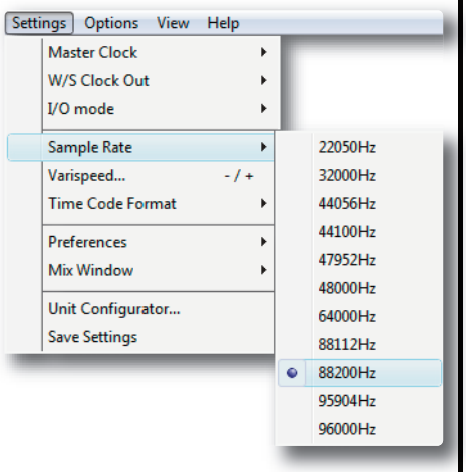
## Additional Sync Out Options

- **LTC**: This setting should be selected to use SSL Soundscape as an LTC Master, with one or more LTC Slave devices synchronised to the Longitudinal Time Code sent via the LTC Out port (RCA/Cinch connector) on the optional Sync Option. (otherwise the LTC option is not available and appears dimmed)
- **BITC**: This setting should be selected to add Burnt-In Time Code to a video signal received via the VITC input on the optional Sync Option (otherwise this option is not available and appears dimmed). The original video signal needs to be fed into the VITC Input (as composite PAL/NTSC Video) and will then be sent with the added Burnt-In Time Code to the VITC output (as a composite PAL/NTSC signal).
- **Midi Clock**: This setting should be selected to use Soundscape as a MIDI Clock Master unit, with one or more Slave devices synchronised to the MIDI Clock and Song Position Pointer messages which are sent via the MIDI Out port when Soundscape is in Play mode or Record mode. The MIDI Clock timing depends on Soundscape's Tempo setting.

*NOTE: The MIDI Clock option is not available when a tempo map is loaded, as described in 'Tempo maps' in the 'File Menu' section of this chapter.*

## Sample Rate

The Sample Rate submenu lists all available clock speeds when the Unit is internally clocked. This menu appears dimmed and cannot be used if the SSL Device (or devices) are clocked to an external device.




Internal Sample Rates

- **22.500Hz** (Multimedia)
- **32.000Hz** (Video and Broadcast Standard)
- **44.056Hz** (NTSC Drop Frame Pull Down of 44.1kHz)
- **44.100Hz** (CD Standard)
- **47.952Hz** (NTSC Drop Frame Pull Down of 48kHz)
- **48.000Hz** (Broadcast Standard)
- **64.000Hz** (Video)
- **88.112Hz** (HD NTSC Drop Frame Pull Down of 88.2kHz)
- **88.200Hz** (High Definition Audio)
- **95.904Hz** (NTSC Drop Frame Pull Down of 96kHz)
- **96.000Hz** (High Definition Audio and Video)

The Sample Rate is also displayed and can be set in the Status Bar, unless an external Master Clock source is selected.

The Sample Rate is shown in red in the Status Bar when Varispeed is active (please read below).

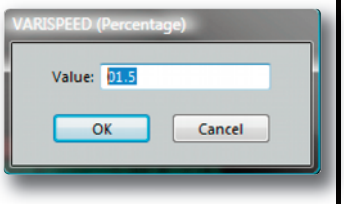



## Varispeed

The Varispeed setting provides a ±10% variation in the Sample Rate. Clicking **Varispeed** opens the Varispeed dialogue where the required value can be entered (as a percentage):




The Varispeed setting can also be adjusted by using the **–** and **+** keys (from the main key area of the computer keyboard only, not from the numeric keypad), which change the varispeed value in –0.1% or +0.1% increments, respectively.

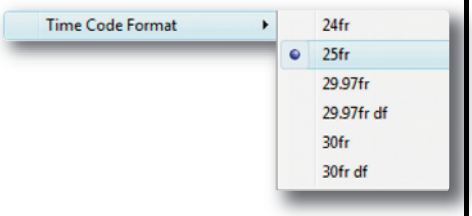



The Varispeed setting is saved with the Arrangement.

*IMPORTANT: Varispeed can be active in RECORD mode. Therefore, if you save an Arrangement with a non-zero Varispeed setting, subsequent recordings within this Arrangement will be made at a non-standard Sample Rate! If the Varispeed value is not 0.00%, the Sample Rate will be shown in RED in the Status Bar. DAT machines and other digital devices may not be able to lock onto the Master Clock signal supplied via Soundscape's audio outputs when varispeed is used.*

## Time Code Format

The Time Code Format selects a Frame Rate commonly used as SMPTE or EBU Frame Rates, such as 24 frames per second for film, 25, 29.97, 29.97df, 30 or 30df frames per second for video. It is required for Synchronisation (external Timecode or APP) and while working with and for Video.

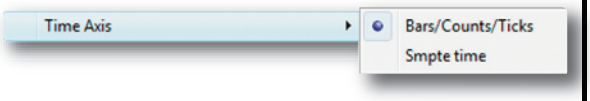



This parameter can also be set in the Status Bar's **FRate** box.

*Note that 29.97 frame rates listed below are in fact the more accurate 30x1000/1001 calculations*

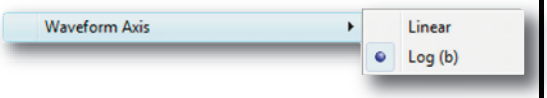
## Time Axis

You can either select musical time divisions (bars/counts/ticks or B/C/T) or SMPTE based time divisions for the Time Axis which is located at the top of the Arrange window.




## Waveform Axis

This parameter defines the mode in which amplitudes in waveforms, automation data and fade curves will be displayed in the Arrangement.

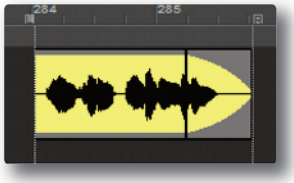



Compared to **Linear** mode, Logarithmic mode (**Log (b)**) visually magnifies lower level details, leaving less visible space to differentiate higher levels.

Logarithmic Mode will also cause a linear fade to appear curved.




**Linear Waveform Axis**

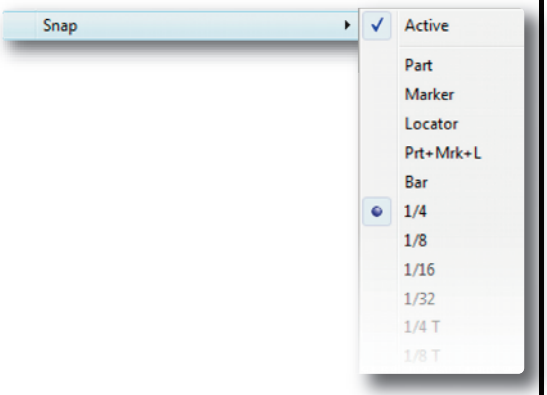



**Logarithmic Waveform Axis**

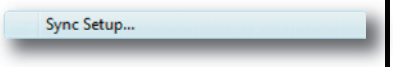
## Snap

The **Snap** function has been described extensively in Chapter 4: *'Status Bar'*.

*The **Snap Point Edit** tool section of Chapter 4: 'Arrange Window': 'Editing Tools' also contains information relevant to the **Snap** function.*




## Sync Setup




### MTC Sync In

When the **Auto Play/Stop when slave syncing** box is checked, Soundscape will always respond to incoming Time Code when an external Sync Source is selected **Settings** / **Sync In**). If the box is not checked, Soundscape will only respond to the selected incoming Time Code when it is in Play mode or Record mode.

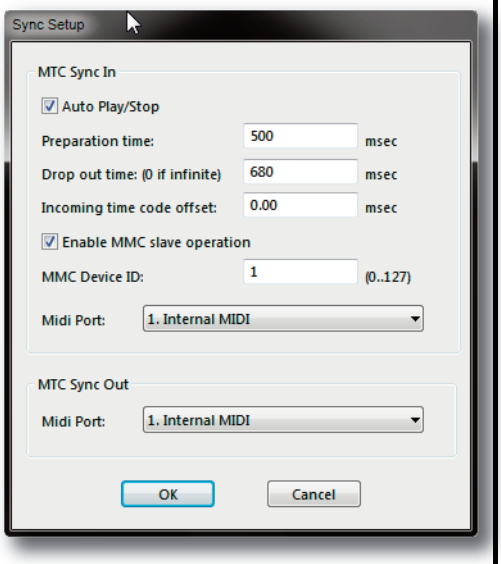
A time value between 100ms and 4000ms can be entered for the Sync in **Preparation Time**. This determines the amount of time which is allowed for filling the internal RAM buffers before Soundscape starts locking to the external Time Code.

The default is 1500ms. It may be possible to reduce this lock time if required, depending on the number of active tracks in the Arrangement and the Disk and overall PC performance.

Please also read the *Recording while slaving to Time Code* subsection of Chapter 4, especially if you need to record from the beginning of an Arrangement while Soundscape is slaved to external Time Code.

A time value up to 21768ms can be entered for the **Drop Out Time**. When Soundscape is slaved to incoming Time Code, this determines how long it will remain in Play mode after the master device has ceased sending proper Time Code or has been stopped. Entering 0 sets the drop out time to infinite. Since Soundscape can not differentiate between Time Code 'Drop Outs' and no Time Code at all (eg. if the Master stops) this free-wheeling time should be set as low as possible.

You can introduce an offset into the incoming time code using the **Incoming time code offset:** box.




It is also possible to enable MMC (Midi Machine Control) Transport Slave Operation, in order to control Soundscape's Transport remotely from a Sequencer or Hardware surface which supports MMC. To do this, check the **Enable MMC slave operation** box, then define the remote device's **MMC Device ID** and the **Midi Port** being used.

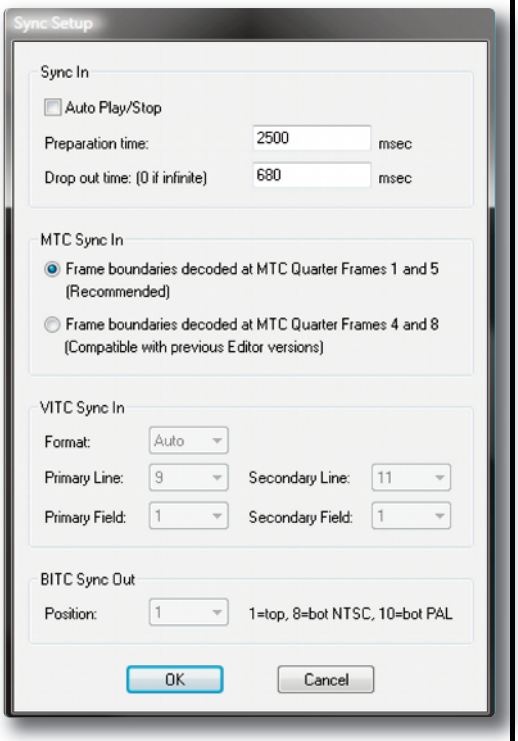
### MTC Sync Out

In the **MTC Sync Out** section, the **Midi Port** on which MTC is transmitted from Soundscape can be defined.

Once you have completed your Sync Setup, click **OK** to save and close the display, or **Cancel** to close the display without saving.

## Legacy Soundscape 32 Sync Options

If you are running a SS32 system, further options will be available in the **Sync Setup** window:




Two radio buttons allow you to select how MTC should be decoded at MIDI Time Code frame boundaries. The first option is fully compatible with the MIDI specification, while the second option should be selected when loading projects started in earlier versions of Soundscape.

### VITC Sync In

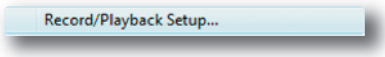
In the **VITC Sync In** section, the Format box can be used to select a VITC format. The Line and Field settings determine the location in the video signal of the VITC information used to decode the VITC data. If the data in the primary location is corrupt, the secondary location is checked. Clicking the arrow to the right of a box opens the list of different value that can be selected.

### BITC Sync Out

In the **BITC Sync Out** section, the on-screen position of the Burnt In Time Code display can be selected.

# Record/Playback Setup

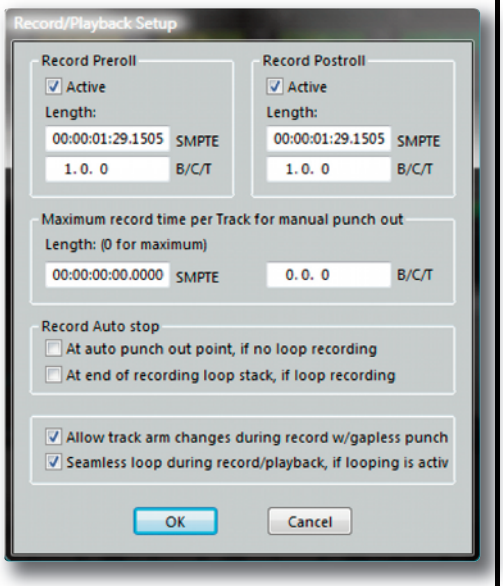
Click to open the **Record Setup** dialogue box.




## Record Pre/Post Roll

The preroll and postroll options can be activated and Pre/Post Roll values can be entered either in SMPTE time or in Bars / Counts / Ticks.

## Maximum record time per Track for manual punch out




You can enter a max. time in SMPTE time or in bars/counts/ticks in the middle section of the window after which Soundscape will drop out of Record mode automatically.

The setting must be '0' if you want to use as much contiguous space as is available on the target drive.

Limiting the potential recording time can improve Disk Performance, since Soundscape will create an empty file for each Recording Track in the set length at the beginning of the recording, potentially decreasing random seek times of the drives due to predictable data locations.

## Record Auto Stop

- If **At auto punch out point (if no loop recording)** is checked, Soundscape will automatically stop upon reaching the Right Locator (or the end of the postroll if active) when recording with Auto Punch Out.
- If **At end of recording loop stack (if loop recording)** is checked, Soundscape will automatically stop upon reaching the Right Locator (or the end of the postroll if active) after the last Take in a record loop stack has been recorded. If several record loop stacks are running concurrently, 'Auto stop' happens at the end of the Stack with the highest Track count.

In the bottom section of the window, checking or unchecking the first box will respectively enable or disable the **Allow track arm changes during record w/gapless punch-out** option. When enabled:

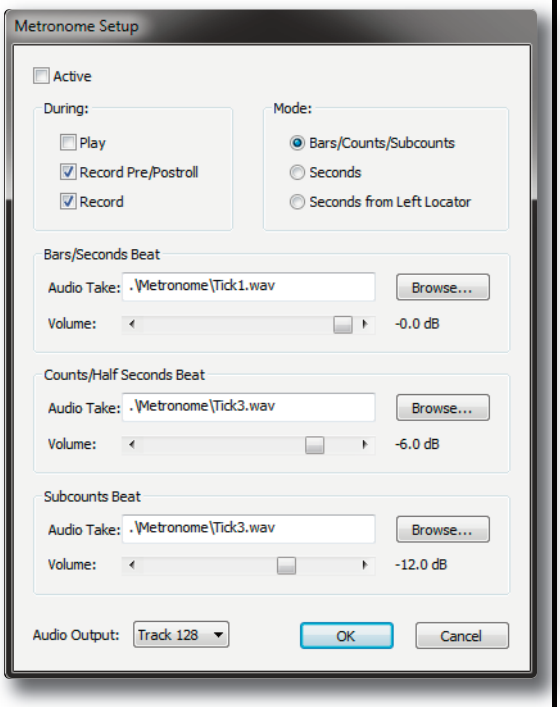
- The track arming buttons can be used in Record mode to perform punch ins and punch outs for individual tracks (the corresponding key commands can be used as well)
- When punching out on audio tracks, the audio after the punch out point is played back immediately, virtually previewing an overdub immediately (à la Tape). The record tracks use double Disk and Buffer bandwidth to provide the gapless punch outs, since Soundscape needs to record a new signal and play back (muted until punching out) the underlying existing Tracks simultaneously.

If the **Seamless loop during record/playback, if looping is active** box is checked, audio is played continuously when the Current Locator jumps from the Right Locator to the Left Locator in Loop mode. If the box is not checked, there is a gap in the audio.

- Seamless looping is ideal for certain tasks, such as practising an instrumental part over a repetitive loop. However, please note that the SSL Soundscape external Time Code Generator does not follow the loop. If it did, the behaviour of a Time Code Slave device (eg., an external Mixing Console with Time Code based automation) would be erratic, since this device would in effect receive 'jumping' Time Code.
- Seamless Looping as an ASIO Positioning protocol Master, however, works in combination with most APP aware applications, if the slaving Sequencer (ie. Nuendo/Cubase) is also in Loop Playback, set to the same Loop I/O locations.

## Metronome Setup

The **Metronome Setup** entry opens a display in which the metronome can be configured:




The **Active** box at the top of the display duplicates the activation icon in the Tape Transport Bar – put a tick in the box to activate the metronome.

The three **During:** options define whether or not the metronome sounds during **Play**, **RecordPre/Postroll** and **Record**.

You can also set the time **Mode** to **Bars/Counts/Subcounts**, **Seconds** (from the start of the Arrangement) or **Seconds from Left Locator**.

You then need to select which audio file is used for each element of the metronome (the **Bars/Seconds Beat**, **Counts/Half Seconds Beat**, and **Subcounts Beat**). To select a file for an element, click on its **Browse** button to open a standard Windows navigation display. Locate the file and click **OK** – the file path will be shown in the **Audio Take:** box.

*Note that default metronome sounds are copied to your hard drive during Soundscape V7 installation.*

You can then define the **Volume** for each beat, using the slider beneath each **Audio Take:** box.

*Note that the audio files used for the metronome must be mono .ATAK or .WAV files*

You also need to assign the metronome an audio track. To do this, click on the **Audio Output:** button at the base of the window, and select a track from the scrollable drop-down list which appears. The selected Track is shown in the **Audio Output:** button.

*Note that this track will be 'stolen' from the Arrangment whenever the metronome is active.*

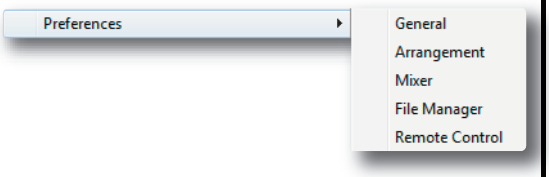
*See the Metronome section of Chapter 4 for more on working with the Soundscape metronome.*

## Automation Setup

The **Automation Setup** entry opens the **Automation Setup** window. All settings in that window have been described in detail in Chapter 4: '*Mixer Automation*'. Please refer to that section for details.

## Preferences

The **Preferences** section provides access to various settings affecting the general behaviour of the software and the way it interacts with hardware.




Five submenus help to organise all available options.

### General preferences




When **Deglitch** is active, each time the Current Locator crosses a Part boundary, Soundscape performs an automatic glitch removal algorithm. This removes the nasty clicks that are sometimes generated when editing digital audio and the amplitude of the waveform did not match a zero crossing (=silence) at the Part boundary.

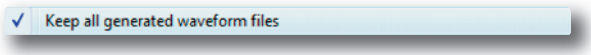
This function effectively removes any need to think about precisely where you cut audio in Soundscape or the necessity to add tiny crossfades to any edit afterwards.

The Deglitch algorithm creates very small fades for you automatically and they are only as long, as they need to be (to avoid clicks).

Soundscape detects audio material that is continuous across a cut (eg., when a cut has been made in order to fade a section of a Part), and does not perform a deglitch in this case.




When the **Auto generate waveform files** option is selected, as soon as a new audio Take is created (by recording or processing), a corresponding waveform file is generated automatically.




When the **Keep all generated waveform files** option is selected, any waveform files that have been generated are saved on the PC's hard disk upon exiting Soundscape. Otherwise, only the waveform files for Takes that are still on the Domain Folders will be saved and all others will be deleted from the PC's hard disk.



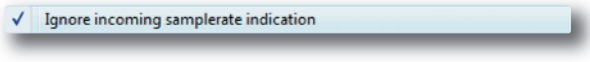

When the **Play till next Marker** option is selected, Soundscape stops playback at each Marker. This is useful for 'previewing' edits, or when 'cueing' up audio for radio spots or theatre. Putting a Marker at the beginning of each audio segment defines 'cue points'. When playback is triggered, it carries on 'till the next Marker' is reached, stopped at the next 'cue point', ready for the next trigger.



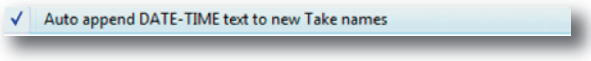

When the **Fast REW / FF** option is selected, rewinding and fast forwarding are performed at four times the default speed. This setting has no effect if the **Scrub during REW / FF** option described below is also enabled.



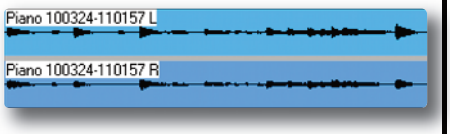

When the **Scrub during REW / FF** option is selected (ticked), audio is played during rewind and fast forward operations. Reverse Play can also be performed at normal playback speed. Please read Chapter 4: *'Tape Transport Bar'* for further details.




When the **Ignore incoming samplerate indication** option is selected, if the Master Clock source for a Soundscape 32 unit is set to AES/EBU (S/PDIF) or TDIF, the incoming AES/EBU or TDIF Sample Rate indication is ignored, and the current Sample Rate is maintained. This is useful if the Master Clock source being used transmits an incorrect Sample Rate indication.



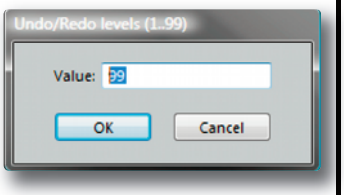

When **Auto append DATE-TIME...** is selected, recorded Takes get the current date and time appended to the end of their name in Year/Month/Day – Hour/Minute/Second format.







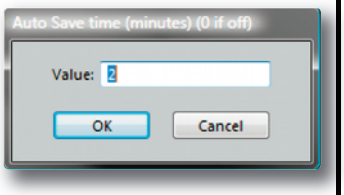
Clicking **Undo/Redo levels** opens a dialogue box where any value between 1 and 99 can be entered to determine the number of available Undo/Redo levels.




*Undo and Redo commands are found in the **Edit** menu. Standard **Ctrl+Z** and **Ctrl+Y** commands can also be used.*




The **Auto Save time** function backs up the current Arrangement and Mixer files automatically and periodically according to a user-defined time interval. This Files are saved in the Record/Process folder marked for the currently open Arrangement.




Clicking **Auto Save time** calls up a dialogue box where the required time interval can be entered. The interval must be entered in minutes and any value between 0 (Auto Save off) and 60 can be entered.

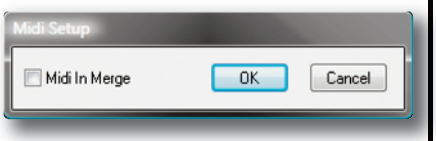



When multichannel audio is recorded as a 4 channel, 6 channel or 8 channel record block, imported or modified, an extension is appended to the Part name. The **Multi-Channel speaker name extension** option can add a **Numeric** extension to the name of each recorded file: **1_x**, **2_x**, **3_x**..., where **x** is the number of multi-channel linked tracks.
If **Surround** is selected, the extensions are: **L**, **R**, **C**, **Cs** for 4 channels, **L**, **R**, **C**, **LFE**, **Ls**, **Rs** for 6 channels or **L**, **R**, **C**, **LFE**, **Ls**, **Rs**, **Lc**, **Rc** for 8 channels.




The **SS32 Midi Port Setup** submenu only shows selectable entries for legacy Soundscape 32 hardware units.

Clicking on the Unit Entry opens a simple dialogue, shown right. If **MIDI In Merge** is active (box checked), Soundscape mixes MIDI received at its MIDI input with Time Code generated internally, and sends the combined data to its MIDI Output. All MIDI messages are merged, except for MIDI 'Real-Time' messages.




A MIDI sequencer could therefore be synchronised to MIDI Time Code or to MIDI Clock and Song Position Pointers generated by Soundscape 32, while MIDI performance data is sent to the sequencer for recording, from a master keyboard via Soundscape 32's Midi Input.

If a sequencer receives a Time Code signal from Soundscape via MIDI while it also has a MIDI output connected to Soundscape's MIDI input, a MIDI loop is created. In this case the MIDI In Merge function should be disabled.

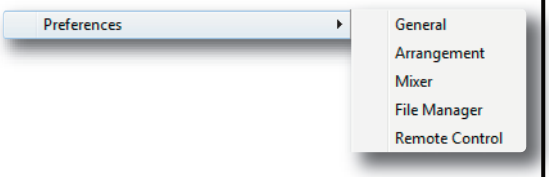



When the mouse pointer is on **AES/EBU (S/PDIF) Xmt Channel Status,** a submenu appears which lets you select **Consumer** or **Professional** format for the digital audio data transmitted from a Soundscape 32's AES/EBU (S/PDIF) outputs.

Changing this setting may be necessary due to the limitations of other devices. For example, some professional DAT recorders will not record from their digital inputs if they receive data in the 'Consumer' format. SSL Soundscape adjusts to either format automatically, therefore there is no equivalent setting for the AES/EBU (S/PDIF) digital input.

## Arrangement Preferences

The **Arrangement** Preferences centralise all options for the visual and logical behaviour of the Arrangement Window. The default settings are chosen carefully. However, when you are experienced with Soundscape, you may want to optimise certain settings for your personal workflow.







While by default the Current Locator moves in the Arrange window according to the current time position, if the **Auto scroll** option is selected, the Current Locator becomes fixed in the middle of the Arrange window, and the Arrangement itself (ie., the Parts, Time Axis, and Marker Bar) scrolls in the relevant direction when Soundscape is in Play, Record, Rewind, or Fast Forward mode.

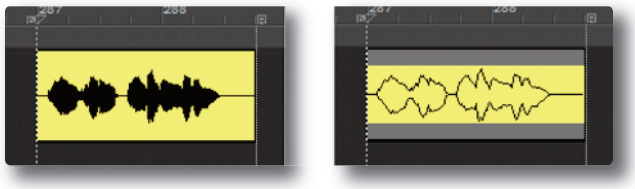
This feature requires a reasonably powerful PC with a fast graphics card, otherwise the movement will be jerky. The smoothness of the scrolling may also be affected by the number of tracks that display waveforms. With **Fill Waveform** turned off (described below), the waveform display will be redrawn faster when scrolling.




Selecting **Fast Redraw** will send the waveform display graphics of each Track directly to the graphics accelerator card. Depending on the speed of the PC and card, this can result in constant, partial screen updates, creating the impression of a 'nervous' screen. When this option is off, the whole screen is updated from memory when necessary, requiring a less powerful graphics card, and in many cases creating a 'smoother' screen redraw.

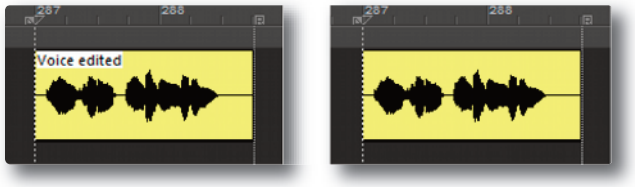



When **Fill waveform** is selected, waveforms are displayed as solid black. Otherwise they are displayed as outlines, which can speed up redraws:



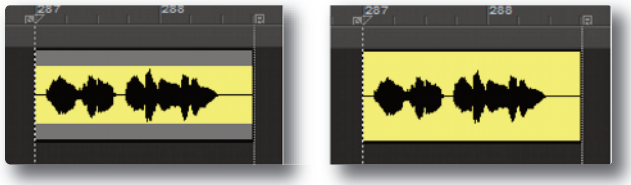



When the **Show Part name in waveform** option is selected, the name of each Part that displays a waveform or automation data is shown. Audio Parts that do not display a waveform do not respond to this setting, as their name is displayed anyway. Waveform based editing may be easier if the Part name is not shown in the waveform.






When the **Show Part volume** option is selected, the volume of each audio Part is represented by the height of the Part's coloured section. If this option is disabled, the volume is still shown for any selected Part(s) or when certain tools (eg., the Volume Trim tool) are used.

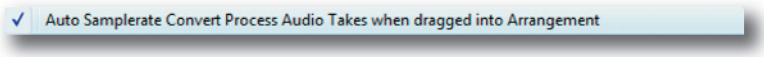





When the **Keep Part selection** option is active, Parts remain selected after they have been edited or processed. This is particularly useful when several editing and/or processing operations must be performed on a group of Parts, and re-selecting these Parts prior to each operation would involve zooming in or out or scrolling repeatedly.

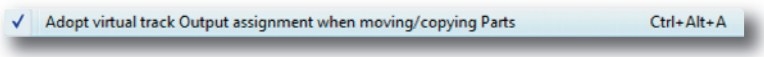



The active/inactive status of the **Shift Markers with Global Insert/Delete** function determines whether any existing Markers will be automatically moved to reflect changes to the Arrangement layout caused by a global 'Cut & Insert' or 'Cut & Delete' operation.



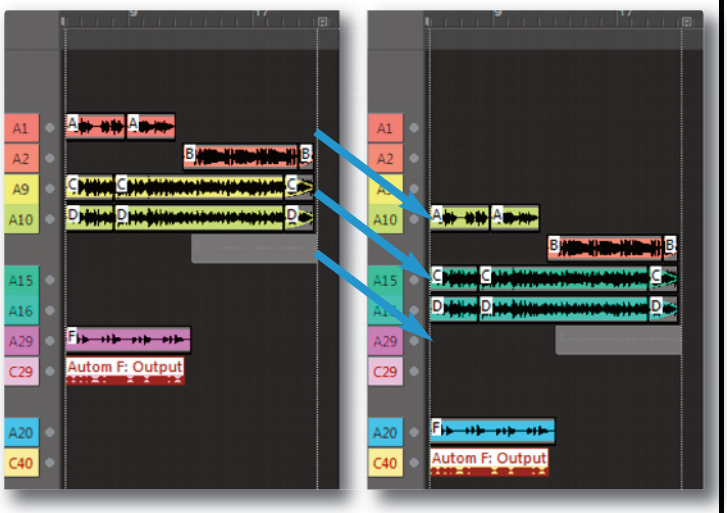

When **Auto Samplerate Convert...** is enabled and a Take is dragged from the File Manager to the Arrange window, the sample rate is converted if necessary to match that of the Arrangement.

*IMPORTANT: As for all Audio Processing tools the source audio Take(s) may be deleted automatically, depending on the setting of the Audio Processing tools behaviour (described in this section below).*

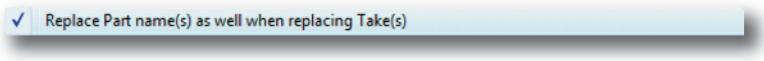



If **Adopt virtual track Output assignment...** is active, any record track inserted in a virtual track slot of the Record Track Column also determines a default output assignment (physical track) for that virtual track. Any Part moved or copied to a virtual track that has an output assignment is automatically assigned to the corresponding physical track. Parts dragged from File Manager will also adopt the Track assignment of the virtual Track they are dropped on.

The group of Parts in the example to the right is being moved down by three virtual tracks using the Move Vertical tool, with '**Adopt virtual track...**' active.




- Audio Part **A** has been moved from track 3 to track 10
  Audio Part **B** ends up on a virtual track that has no output assignment therefore it remains on track 5.
  Audio Parts **C** and **D** (stereo linked, originally assigned to tracks 9-10) have moved to tracks 15-16.
- Track assignments for moved stereo linked Parts will only be changed if both destination virtual tracks have an output assignment. (even if they are mono Tracks with non consecutive numbers).
- Part **E** moves to a virtual track assigned to output track 29, but remains muted.
- Part **F** (originally assigned to track 29) has been assigned to track 20.
- Automation Part **Autom F...** (originally assigned to automation track 29) has been assigned to automation track 40.



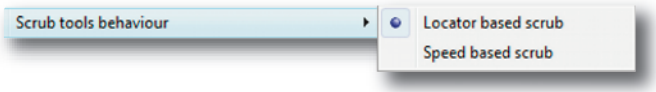

When **Replace Part name(s) as well when replacing Take(s)** is active, every time a Take referenced by a selected audio or automation Part in the Arrange window is replaced by right-clicking an appropriate Take in the **File Manager** and selecting **Replace Take(s)** in the **Context** menu, the name of the selected Part is replaced by the name of the new referenced Take automatically. If multiple Parts have been selected, they are all renamed to match the newly referenced Take.



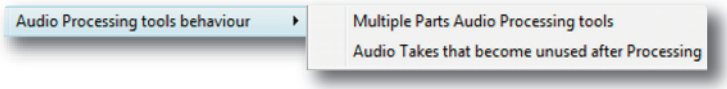

By default, the screen is redrawn to follow the Current Locator whenever it exits the visible section of the Arrangement.

When the mouse pointer is on **Arrange window follows current locator**, a submenu appears to activate or deactivate this function (which can also be toggled using the ***J*** key or the Follow Current Locator button located in the top right corner of the Arrange window).

The **Never during solo** option keeps the Arrange window from following the Current Locator, while the Solo tool is used. This is particularly useful at high zoom levels and while using the Solo tool to check an edit point when you wish to stay in focus.

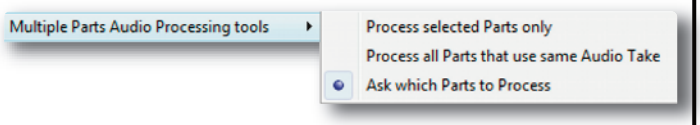



With the **Scrub tools behaviour** option, you can set the Scrub tools to be either Locator or Speed based. The chosen setting applies to the behaviour of the Solo Scrub and Multitrack Scrub tools. You can find detailed information on how to use the Scrub Tools with the different behaviours in Chapter 4: '*Editing Tools*'.

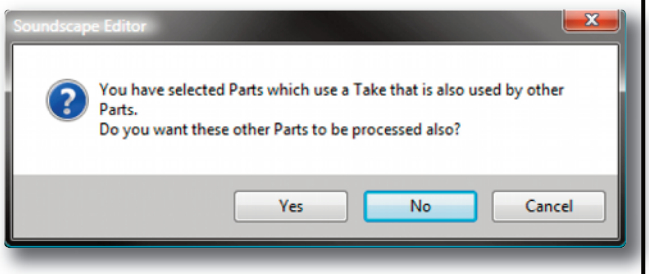



The **Audio Processing tools behaviour** option is divided into a submenu with two entries: **Multiple Parts Audio Processing tools** and **Audio Takes that become unused after Processing**.

In **Multiple Parts Audio Processing tools** you have three further options, defining what happens whenever a multi-Part audio processing tool is used on a Part:

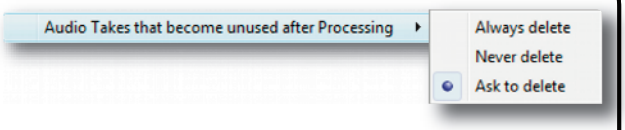



- **Process selected Parts only**: the processing is applied to that Part only. Other Parts that reference the same audio Take are not affected in any way.
- **Process all Parts that use same Audio Take**: the processing is also applied to any other Parts that refer to the same audio Take.
- **Ask which Parts to Process**: the dialogue box shown right appears, letting you specify whether in this particular instance the processing should also be applied to other Parts that refer to the same audio Take.

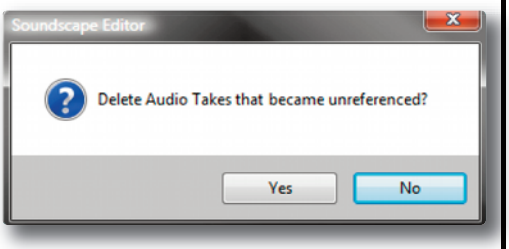



These settings only apply for the **Normalize** and **DC Removal** tools, which currently are the only Multi-Part Audio processing Tools.

In **Audio Takes that become unused after Processing** you again have three options defining what happens whenever an audio Take becomes unused after an audio Part has been processed:

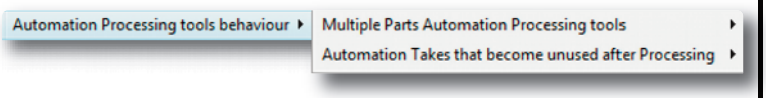



- **Always delete**: it is deleted automatically without notice.
- **Never delete**: no action is taken. The Take remains on disk.
- **Ask to delete**: the dialogue box shown right appears, letting you specify whether in this particular instance the unused (unreferenced) Take should be deleted.

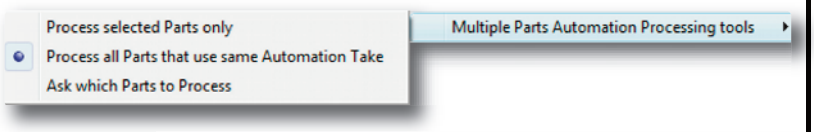



These settings apply for all audio processing tools.

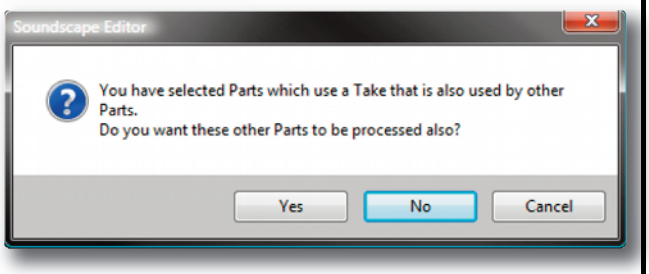
The **Automation Processing tools behaviour** option is divided into a submenu with two entries: **Multiple Parts Automation Processing tools** and **Automation Takes that become unused after Processing**.



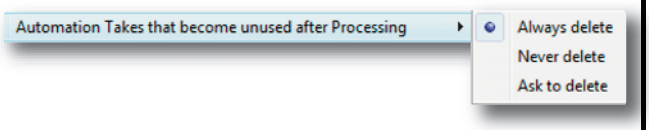

In **Multiple Parts Automation Processing tools** you have three options, defining what happens whenever an automation processing tool is used on a Part:



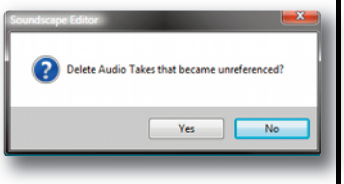

- **Process selected Parts only**: the processing is applied to that Part only. Other Parts that reference the same automation Take are not affected in any way.
- **Process all Parts that use same Automation Take**: the processing is also applied to any other Parts that refer to the same automation Take.
- **Ask which Parts to Process**: the dialogue box shown right will appear, letting you specify whether in this particular instance the processing should also be applied to other Parts that refer to the same audio Take.



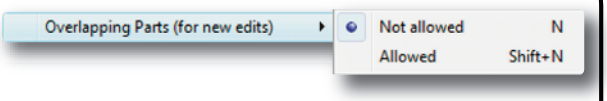

In **Automation Takes that become unused after Processing** you again have three options, defining what happens whenever an automation Take becomes unused after an automation Part has been processed:




- **Always delete**: it is deleted automatically without notice.
- **Never delete**: no action is taken. The Take remains on the disk.
- **Ask to delete**: the dialogue box shown right appears, letting you specify whether in this particular instance the unused (unreferenced) Take should be deleted.



The **Overlapping Parts (for new edits)** option determines whether Parts in the Arrange window can overlap one another.




You can also type ***Shift+N*** or ***N*** to allow or not allow Part overlaps respectively.

When Parts (or sections of Parts) overlap, any hidden Parts (or sections) are muted while visible Parts plays back normally: 'what you see is what you hear'.

***Overlaps are always allowed for crossfades (created using the Crossfade tool) regardless of the current setting.***

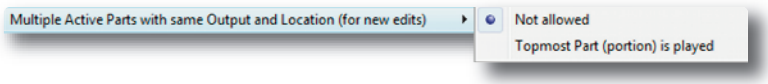
If you select **Not allowed**, previously existing overlaps are preserved. An overlap that has been suppressed (eg., by moving a Part) can be restored by using the Undo function, even if **Not allowed** has been selected.

The Parts layering system is organised in a z-order scheme. By default, any Part that has never been overlapping or overlapped by another Part is in layer 0 of the z-order.

Assuming overlaps are allowed as described above:

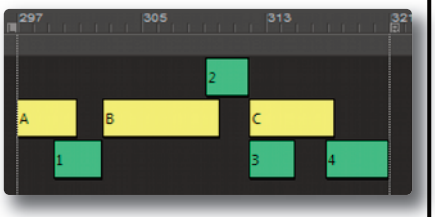
- When a Part is edited in such a way that an overlap is created between that Part and another Part in the same layer and on the same virtual track (eg., using the Move tool, Copy tool, or Trim tool), the edited Part gets moved up to the next z-order layer.
- When a Part is edited in such a way that overlaps are created simultaneously between that Part and several other Parts in different layers on the same virtual track, and if the edited Part is in the same layer as one of the overlapped Parts, the edited Part is shifted up in the z-order so that it is in the next layer up from the overlapped Part(s) in the highest numbered layer. For example, if 'Part D' (layer 0) is moved so that it overlaps 'Part A' (layer 2) and 'Part C' (layer 0), 'Part D' is shifted up to layer 3.
- When a Part is edited in such a way that an overlap is created between that Part and another Part in a different layer on the same virtual track, the Part assigned to the highest numbered layer ends up above the other Part. For example, if an overlap is created between 'Part D' (layer 3) and 'Part E' (layer 1), 'Part D' will end up above 'Part E', regardless of which one was edited, because this is implied by their existing z-order layer assignments.
- When a Part is edited in such a way that an overlap is created between that Part and another Part which was on a different virtual track prior to the edit (eg., using the Move tool or Copy tool), the edited Part is shifted up in the z-order if necessary so that it is in the layer above the overlapped Part. For example, if 'Part A' (layer 2 in virtual track 7) is dragged onto 'Part F' (layer 3 in virtual track 9), 'Part A' is shifted up to layer 4.

*NOTES:*
*In practice, if you want a Part to be on top of another and find that it slots underneath that other Part, all you need to do is drop it onto another virtual track, then move it back where you want it. It will then end up above the other Part.*
*If an overlapped Part is completely hidden from view, selecting the Part(s) above it for editing by drawing a selection box may include it as well (the hidden Part gets selected if it is included in the selection box, which cannot be visually checked). To make sure that hidden Parts are not edited by mistake, it is preferable to edit the visible Parts one by one, or to hold the **Ctrl** key and click the required Parts to select them.*
*If overlaps are not allowed, recording over existing Parts or sections of Parts causes these Parts (or sections) to be removed from the Arrange window. If overlaps are allowed the new Parts are stacked on top of the existing ones.*

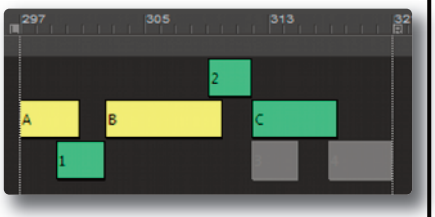



The **Multiple Active Parts with same Output and Location (for new edits)** entry lets you determine whether multiple active Parts that overlap in time (regardless of their virtual track position) can have the same output assignment (ie., track assignment).

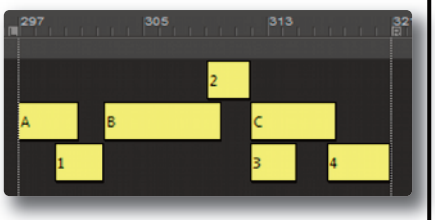
If **Not allowed** is selected, whenever an active Part is given a new track assignment, any Parts already assigned to that track which overlap the edited Part in time become muted. In the example, Parts A, B, and C are assigned to track 8, (colour-coded yellow), and Parts 1, 2, 3, and 4 are assigned to track 11 (colour-coded green).




The Track Assign tool is then used to change Part C's track assignment from track 8 to track 11. Part 3 and Part 4 are muted as a result.




If **Topmost Part (portion) is played** is selected, any active Part can be assigned to any track without affecting the mute status of any other active Parts already assigned to that track. It is therefore possible to have overlapping active Parts with the same track assignment. If this is the case, in Play mode, Soundscape only plays the topmost Part (or portion) at any given time position. In the example on the right, all the Parts are assigned to the same track.



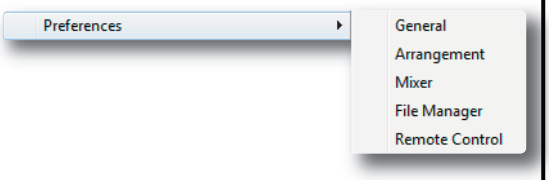

Part A will be played from bar 297 to bar 301 included, Part 1 will be played only for one bar (bar 301-302), Part B will be played from bar 302 to bar 309 included, Part 2 will be played from bar 309 to bar 311 included, Part C will be played from bar 312 to bar 315 included, Part 3 will not be played at all, and Part 4 will be played from 316 on.

If the setting is changed from **Topmost Part (portion) is played** to **Not allowed**, active Parts in the same time range which already have the same track assignment all remain active (hence the mention 'for new edits'). In Play mode, at any point in the timeline, only the topmost of these Parts (or topmost portion) is played. If one such Part is then muted or reassigned to a different track, it can still be restored to its former state by using the 'Undo' function.

*NOTES:*
*This setting applies to audio and automation Parts.*
*When **Topmost Part (portion) is played** is selected, depending on the current zoom level and scroll box positions, a Part outside the visible portion of the Arrangement may be played instead of a visible Part. In particular, when editing at a high zoom level, the Part being edited may not be heard in Play mode. It is therefore preferable in this case to use the Solo tool to check an edit.*

## Mixer Preferences

The **Mixer** Preferences include all options for the visual and logical behaviour of the Mixer Window.



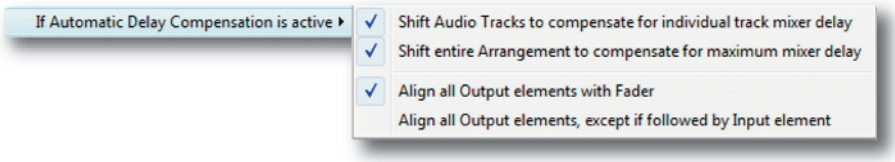

The default settings are chosen carefully, however, certain work scenarios may require different settings.




When **Auto route** is active, the DSP resources are allocated automatically to the various mixer elements, including plug-in effects, etc. When the option is inactive, these resources can be allocated manually by selecting the DSP cells for any Mixer Column upon creation. Mixer Columns can be manually assigned, even if this option is active, by clicking into an empty Mixer column area with the I/O Assign Tool.

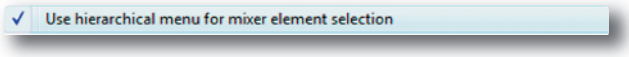



When **Keep Column Selection** is selected, a multiple column selection will remain selected until a new selection is made or an empty column is selected, rather than deselecting once a tool has been used.

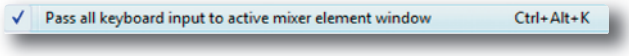



The **If Automatic Delay Compensation is active** heading opens a further submenu, giving access to several optional settings relating to the operation of the Automatic Delay Compensation (ADC).

Please refer to Chapter 4: '*Mix Window*': '*Automatic Delay Compensation (ADC)*' for more details.

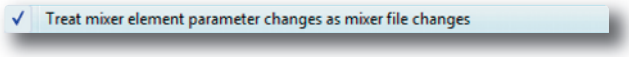



When the Create tool is used in **Mix Edit** mode to insert a mixer element into a mixer column, a menu appears that lists all available mixer elements. The size of this menu can make it difficult to find a particular element, especially if a lot of plug-ins are installed or if a small monitor is used. If the **Use hierarchical menu for mixer element selection** option is enabled, the menu is organised into several sections and pop-up submenus.

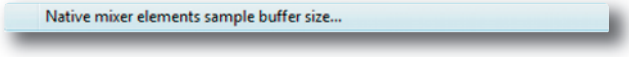



When **Pass all keyboard input to active mixer element window** is enabled, input from the computer keyboard is passed to the active mixer element or Plug-In window. This may be necessary if the Plug-In provides a special set of Shortcuts or when you need to type in some codes to authorise a VST plug-in.

*Note that in this case, the keyboard shortcuts for Soundscape may cease to work properly.*




When **Treat mixer element parameter changes as mixer file changes** is enabled, a change to a mixer element parameter will enable the **Save Mix** function under the **File** menu. If it is disabled, only changes to the structure or routing of the Mixer enable the **Save Mix** function.

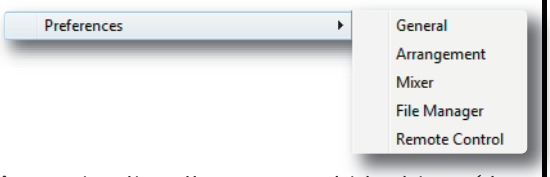


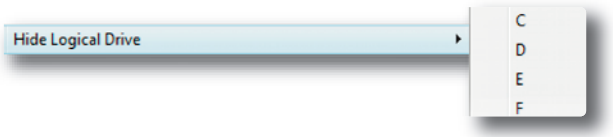

Clicking on **Native mixer elements sample buffer size** opens a dialogue box where a value in samples can be entered for the native mixer elements buffer size. The default value is 128. Typically, it should be raised if the PC is struggling to cope with the processing demands.

Please refer to Chapter 3: '*Create your first Soundscape Project*' for more details.

## File Manager Preferences

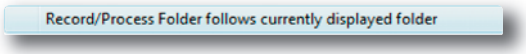
The **File Manager** Preferences has two sub-menus:




The **Hide Logical Drive** option list allows you to hide drives (that have a Drive Letter) from the Drive Bar of the File Manager window. This is especially useful, when you want to prevent Read Only Drives (CD/DVD-Rom), Networked Drives , Back-Up Storage or the System Drive to be accidentally used in a Soundscape recording session.

Selecting/ticking a drive will hide it. You can select Drive Letters from A: to Z:, even if those Driver Letters are currently not assigned.



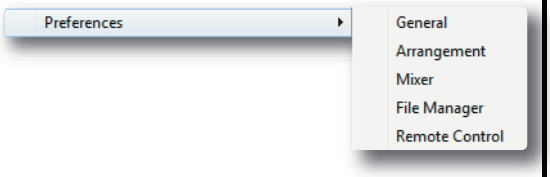

When **Record/Process Folder follows currently displayed folder** is active, Soundscape records or processes new audio and automation Takes to the currently displayed folder inside the File Manager Window.

This option removes the need to map a Record/Process Folder in any new Arrangement. It resembles the SDisk Folder behaviour of legacy Soundscape Editor Versions (V5.5 and earlier).

*NOTES:*
*Enabling this option can quickly spread the Takes necessary for an Arrangement across multiple folders.*
*Autosave files of the current arrangement and mix are still saved to their original folders.*

## Remote Control Preferences

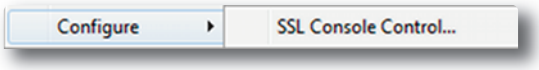
The **Remote Control** preferences includes two sub-menus, related to control of Soundscape from an external control surface.







Selecting **Connect to** opens a further submenu, where Soundscape can be connected to either the **Legacy Console Manager** or **SSL Console Control** application. To run Soundscape with no external control, deselect the active connection.



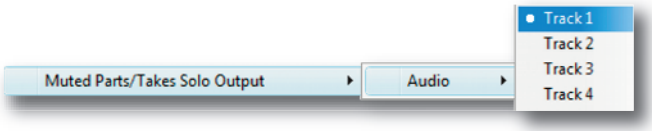

When connected to SSL Console Control, selecting **Configure** / **SSL Console Control...** opens the **SSL Console Control** window for configuration

*Please see the 'Console Control' section of this guide for more details about using either the Legacy Console Manager or SSL Console Control.*

# Various Settings

The remainder of the **Settings** menu is made up of the following options:

## Muted Parts Solo Output

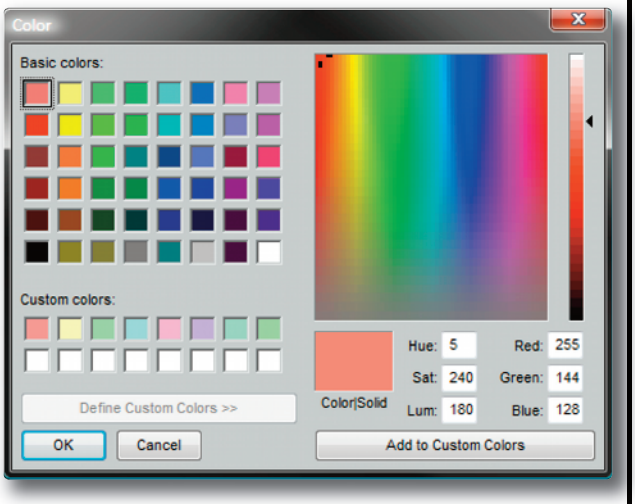



Muted Parts in the Arrangement, and audio Takes in the File Manager, can be played back or scrubbed through the Soundscape Mixer using the Solo and Scrub Tools. When you click on **Muted Parts/Takes Solo Output → Audio**, submenus let you specify, for each Unit in your system, which physical track should be used for playback in this instance. This does not affect the Arrangement track assignments – the chosen track is just borrowed when the Solo tool is used.

*NOTE: When muted stereo-linked Parts are played back, the odd/even numbered track pair which includes the Solo Output track selected here is used automatically.*

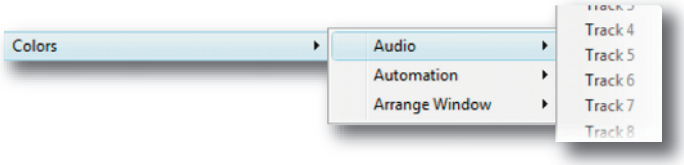
## Colours

If you click on **Colours**, several submenus appear which can be used to access standard Windows colour palettes and customise the look of certain elements of Soundscape.




*Since it can take quite some time to define colours in larger sessions, remember to click **Save Settings** in the **Settings** Menu!*

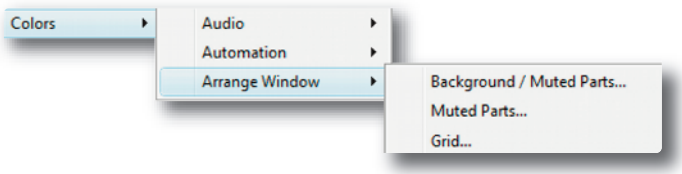
The colour setting of any audio Track can be accessed and edited by clicking the **Track** Number in the **Audio** sub menu:




*NOTE: For better clarity while editing visually the Waveform Displays will always remain black (filled or as an outline). Hence Colours with a Luminosity of less than 128 (50% Brightness) may become too dark to differentiate the Track background colour from the waveform, possibly straining your eyes during longer editing sessions.*

Similarly, the colour setting of any automation Track can be accessed and edited by clicking the **Track** Number in the **Automation** sub menu.

*NOTE: Unlike Waveform colours, Automation colours adapt to changes in Automation Track background colours. The underlying algorithm selects the greatest Contrast available within a similar colour range. For any background colour with a luminosity of around 128 (50%) the algorithm can not find a matching colour with a high contrast. It is therefore better to use light or dark colours for the background.*

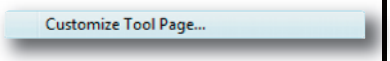



The colour setting of the Arrange window background and muted Parts (or muted Parts on their own), and Grid can be accessed and edited by clicking the corresponding item inside the Arrange Window submenu.

*TIP: Try to avoid extreme contrast for 'background information' (ie. the Grid) or any extremely saturated colour for the background. The main information inside the Arrange Window are Takes, Waveforms and automation curves, and those should visually be in the 'foreground'.*

## Customize Tool Page

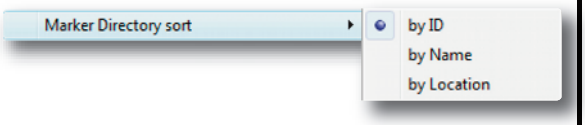
**Customize Tool Page** opens the identically named Dialogue Box to customise and re-arrange the currently displayed Tools in the Toolbar.




Please refer to Chapter 4: '*Toolbar*' for a detailed explanation.

## Marker Directory sort

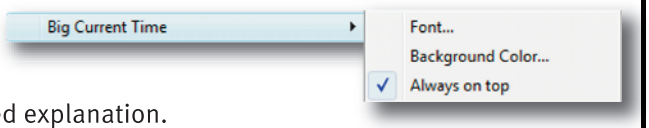
**Marker Directory sort** allows you to choose one of three display options for sorting in the Marker Directory Window:




- **by ID**: Markers are ordered according to their ID number (lowest number first)
- **by Name**: Markers are displayed in alphabetical order, if you have named them. Markers that have the same name (or none) are sorted by their time position.
- **by Location**: Markers are ordered according to their time position in the Arrangement.

## Big Current Time

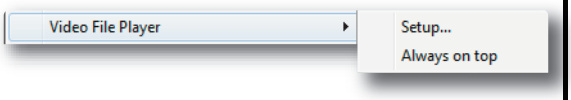
**Big Current Time** opens a submenu with different entries to customise the appearance of the Big Current Time window.




Please refer to Chapter 4: '*Big Current Time Window*' for a detailed explanation.

## Video File Player

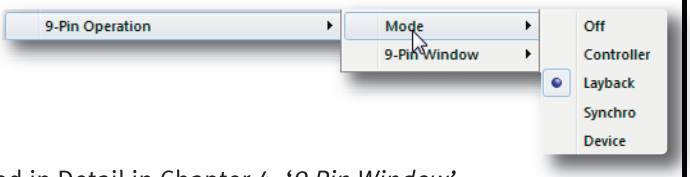
**Video File Player** opens a submenu which lets you access and define settings for Soundscape's integrated Video File Player.




Please refer to Chapter 4: '*Video File Player Window*' for a detailed explanation.

## 9-Pin Operation

The **9-Pin Operation** item is only available if a Soundscape 32 with an installed Sync Option is part of the current System Configuration. The item appears greyed out if this is not the case.




The functionality and Setup of the 9-Pin Operation is described in Detail in Chapter 4: '*9 Pin Window*'.

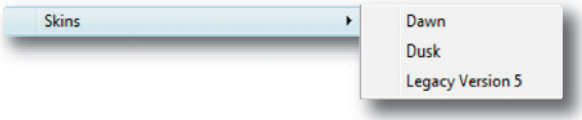
## Mix Window




When the mouse pointer is on **Mix Window**, a submenu appears that lets you activate or deactivate the **Always on top** option.

If selected, the Mixer window will always remain visible on top of any other window, even if that other window is active. The exception to this is if that other window is also set to be **Always on top**: the **9-Pin Master**, **Big Current Time**, and **Video File Player** windows also have this option.

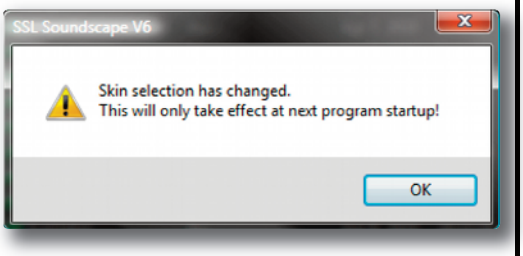
## Skins



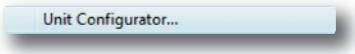

With the **Skins** option you may chose between different Soundscape looks.

The skin selection changes most of the Arrange, File Manager and Main Window appearance, while keeping the Audio and Automation track colour settings intact.

After selecting a skin, a message (shown right) informs you that your selection will take effect at the next program startup. Click **OK** to continue.




## Unit Configurator




Clicking on **Unit Configurator** opens the **Unit Configurator** window which can be used to change the Hardware System Configuration, including all installed SSL MX4 or Mixpander units (and under certain circumstances, legacy Soundscape Systems also).

This utility is described in Chapter 3: '*Unit Configurator*'.

## Save Settings



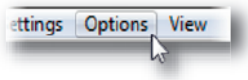

Clicking **Save Settings** saves the contents of the Soundscape **Settings** menu.

This also includes the current selection of Editing tools, all Windows Sizes and Positions, the vertical and horizontal zoom levels as future default values.

***Any changes you make to the global Settings are temporary and will be reset to previous defaults when re-launching Soundscape, if Settings haven't been saved.***

# Options Menu

Clicking on the **Options** menu opens the **Passwords** window, which lists all included or optional software plug-ins that are password protected and installed on the host computer.

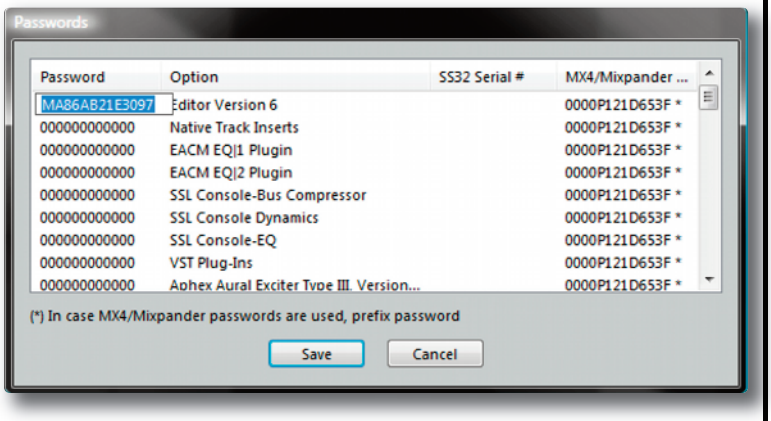



*Please note that the Soundscape Software itself must be enabled by entering the correct password for the installed hardware unit.*

Each SSL hardware unit has a unique serial number inside the DSP Hardware (Unique ID or UID). Authorisation Passwords are also unique and will only enable modules on the individual hardware they were generated for.

Some options have multiple entries in the list; However only one has to be filled with the right password, which will then be automatically copied to the other entries.

- The Mixer Password for a new SSL hardware unit can be found on a sticker that is placed directly on the board or on the product packaging and registration card. It can also be obtained from our website if you have registered the unit.
- For MX4, your sticker should also contain Passwords for the 3 Plug-ins of the SSL Console Bundle and another Password for the SSL Audio Toolbox.
- If you haven't purchased Soundscape V7, you can work with Soundscape V7 without any limitation in functionality (apart from the missing Time Module functions) and without any time limit. However, Soundscape V7 will be limited to the recording and playback of Tracks 1-16. All other Tracks will be greyed out and remain silent (until you purchase the full version!).



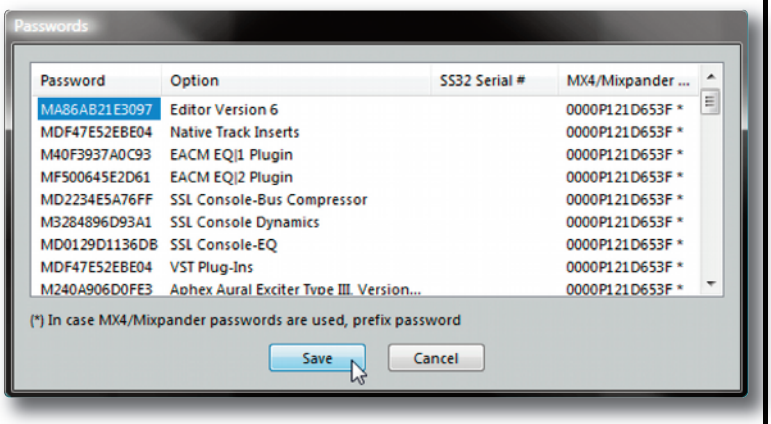

- If you have purchased Soundscape V7 you will have received the Soundscape password along with a password for the XPro Time Module.

Initially, all passwords are set to '000000000000'. When entering a plug-in password generated for an MX4 or Mixpander card, a leading M (capital letter M) needs to be entered.

If several cards are present in the system, make sure that the card's serial number in the right-hand column matches with your password entry.

To enter a password simply double click on the password text field.

When all passwords have been entered, click the **Save** button to enable them and exit the window.

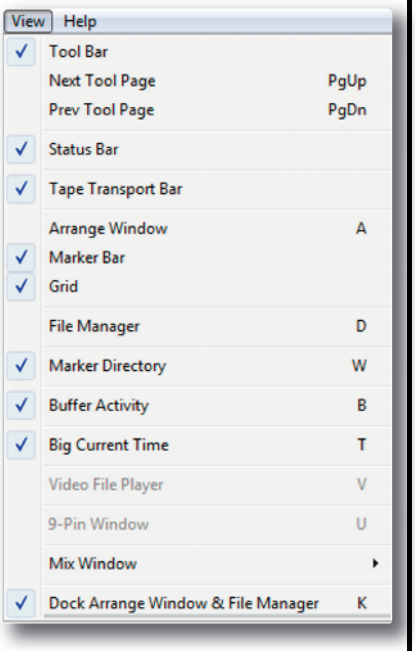



*NOTES:*
*If you have purchased a password for only one unit, this module will be enabled on all the hardware units in the system that run inside the same instance of Soundscape. The password should only be entered for the unit it was generated for, the other unit(s) should be left at the default '000000000000'. Only then Password Sharing will enable this module on all Units.*
*Certain Modules listed in the Passwords Window (Cue Sheet Printing, Standard Module, Pro Tools Import/Export) do NOT need a Password entry.*

# View Menu

## Overview

This menu allows you to configure the appearance and visibility of Soundscape's functional Windows.




The following windows can be resized and positioned anywhere on Windows Desktop, including extended Desktops for Multi-Monitor Setups:

- The **Tool Bar**, **Status Bar** and **Tape Transport Bar** within the Main Soundscape window
- **Arrange Window** with its **Time Axis** and **Marker Bar** ***A***
- **File Manager** window ***D***
- **Marker Directory** window ***W***
- **Big Current Time** window ***T***
- **Video File Player** window ***V***
- **Mix Window** ***Q*** or ***X***

The **Tool Bar**, **Status Bar**, **Tape Transport Bar**, and **Marker Bar** can be shown or hidden by clicking the corresponding menu entry.

The following windows cannot be resized, but can be positioned freely

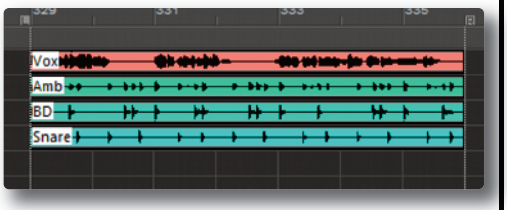
- **Buffer Activity** window ***B***
- **9-Pin Window** ***U***

All screen settings including size and positions can be saved by clicking **Save Settings** in the **Settings** menu.

The **Next Tool Page** and **Prev Tool Page** (***Page Up***/***Page Down***) items can be used to scroll through the Tool Pages.

## Grid

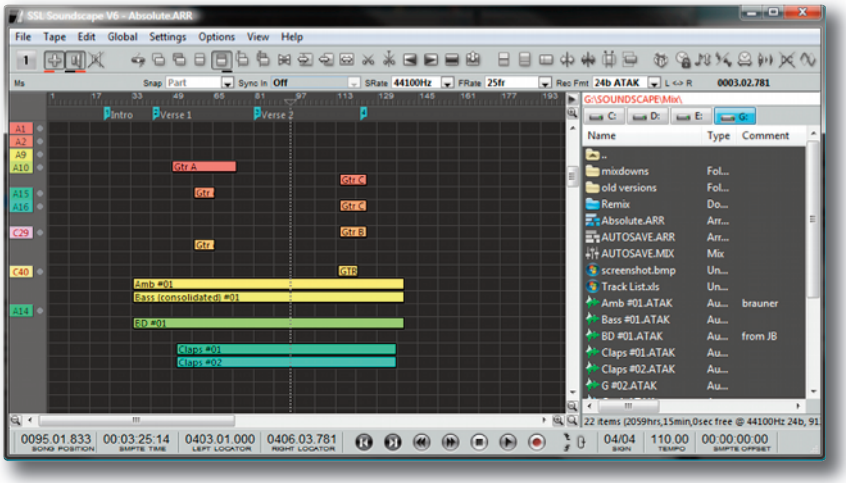
The **Grid** entry shows or hides the Arrange window's grid. The grid is composed of horizontal lines which indicate virtual track positions, and vertical lines which indicate time subdivisions.




If the **Time Axis** is set to **SMPTE** time in the **Settings** menu, the highest possible zoom level will display one vertical line per frame. If the **Time Axis** is set to **bars/counts/ticks**, the highest possible zoom level will display one vertical line for every four ticks (there are 960 ticks per quarter note).

## Dock Arrange window & File Manager

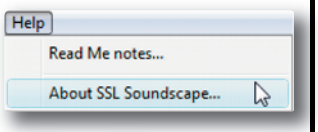
Selecting **Dock Arrange Window & File Manager** docks the Arrange and File Manager windows to the Main window, saving screen real estate and allowing proportional adjustment of both windows when resizing the main window. Sliding the separator line between the Arrange Window and File Manager also makes it easy to gain space in one window while the other one shrinks accordingly.



*NOTE: When this is active, you cannot hide the Arrange or File Manager Window with either Menu entries or keyboard shortcuts.*

# Help Menu

The **Help** menu allows direct access to the Readme Notes, including last version changes, along with the **About SSL Soundscape** Window which displays additional System Information.




All Soundscape related User Guides can also be accessed from this Menu (including this User Guide).

## User Guides

Clicking on any **User Guide** (MX4 hardware installation, Soundscape V7, and SSL Console Bundle) opens it in a PDF Format inside an appropriate reader application.

*Note that you will need to ensure that you have a PDF Reader installed on your System.*

## Read Me notes

The Readme Notes contain the latest changes in software revision. They may contain important last minute changes which have not yet made it into the User Manual.

## About SSL Soundscape

This entry opens a typical **About..** Window, with Software Version Information, some licensing and copyright info and a nice and polished graphic:)




Note that you may frequently need the Information at the bottom of this window, where all currently configured Units are listed with their unique serial number (UID). The UID is required whenever you want to purchase additional software modules or want to register the hardware on our website. In addition, our Service Department or your Dealer/Distributor may want to know this number in case you need any help, so please remember this window.

Click **OK** to close the window.

# 6. Soundscape Keyboard Shortcuts

| File | |
|---|---|
| Open Arr | CTRL+O |
| Append Arr | CTRL+A |
| Save Arr | CTRL+S |
| Save Mix | CTRL+Q |
| Exit | ALT+X |

| Tape | |
|---|---|
| Go to Begin | Home |
| Go to End | End |
| Got to Prev Snap | SHIFT+LEFT |
| Go to Next Snap | SHIFT+RIGHT |
| Go to Prev Part Border | CTRL+LEFT |
| Go to Next Part Border | CTRL+RIGHT |
| Rew | LEFT |
| Fw | RIGHT |
| Stop | DOWN / Spacebar |
| Play | UP / Spacebar |
| Rec | Num ADD |
| Go to Left Loc | ALT+L |
| Go to Right Loc | ALT+R |
| Auto Punch In | CTRL+ALT+I |
| Auto Punch Out | CTRL+ALT+O |
| Loop | Y |
| Arm Audio Trk | NUM MULTIPLY |
| DisArm Audio Trk | NUM DIVIDE |

| View | |
|---|---|
| Show Arrange window | A |
| Show Mix Window | Q (narrow) /X (wide) |
| Up a tool page | Page Up |
| Down a tool page | Page Down |
| Show Marker Directory | W |
| Show File Manager | D |
| DOCK Arr and File Manager | K |
| Time Code Display | T |
| Buffer Activity | B |
| Video Player | V |
| 9-pin Window (SS32+Sync Opt.) | U |

| Global | |
|---|---|
| Cut/Ins L<->R | SHIFT+I |
| Cut/Del L<->R | SHIFT+D |
| Cut/Sel L<->R | SHIFT+C |
| Cut at Current Loc | SHIFT+F |
| Move selct'd Parts to L Loc | SHIFT+Q |
| Copy selct'd Parts to L Loc | SHIFT+W |
| Select all Parts | SHIFT+A |
| Select all P's same virtual TRK | SHIFT+V |
| Select all P's same OUT | SHIFT+O |
| Select all P's same TAKE | SHIFT+T |
| Select all P's Part to End | SHIFT+G |
| Select all muted Parts | SHIFT+M |
| Select Timelocked Parts | SHIFT+K |
| Select P's L<->R Loc | SHIFT+S |
| Select P's L Loc -> End | SHIFT+R |
| Unselect all Parts | SHIFT+U |
| Invert Part selection | SHIFT+X |
| Set L+R Loc start/end sel P's | SHIFT+L |
| Set L Loc sel P's end | SHIFT+E |
| Set R Loc sel P's begin | SHIFT+B |
| Flatten Arrangement | CTRL+SHIFT+K |
| Compute Tempo L<->R | CTRL+SHIFT+J |

| Settings | |
|---|---|
| Sync In = OFF (Master) | M |
| Sync In = MTC | S |
| Sync In = LTC (SS32+Sync Opt.) | O |
| Varispeed Up (+0.1%) | ADD (not NUM pad) |
| Varispeed Down (–0.1%) | MINUS (not NUM pad) |
| Snap Active - toggle | H |
| Follows Current Loc | J |
| O'lap Not Allowed | N |
| O'lap Allowed | SHIFT+N |
| Keep Part Selection | SHIFT+P |
| Auto Generate Waveform | CTRL+SHIFT+W |
| Auto Scroll Arr Window | CTRL+SHIFT+I |
| Adopt Output Assign | CTRL+ALT+A |
| Pass keyboard to mixer elem | CTRL+ALT+K |

| Edit | | | |
|---|---|---|---|
| Create | ALT+1 (number one) | Noise Gate | CTRL+K |
| Track Assign | CTRL+1 (number one) | Waveform | CTRL+W |
| Move | CTRL+M | Info | CTRL+I (letter i) |
| Move Vert | CTRL+SHIFT+M | Delete | CTRL+D / DEL |
| Move Loc | CTRL+ALT+M | Stereo Link | CTRL+L |
| Copy | CTRL+C | Multi-Ch Link | CTRL+4 |
| Copy Vert | CTRL+SHIFT+C | Slip | CTRL+J |
| Copy Loc | CTRL+ALT+C | Invert Phase | CTRL+P |
| Cut | CTRL+X | Repeat | CTRL+R |
| Cut Loc | CTRL+ALT+X | Timelock | CTRL+SHIFT+L |
| Volume Tool | CTRL+V | Auto-Curve Select | CTRL+SHIFT+A |
| Vol Fade In | CTRL+E | Auto Event Edit | CTRL+SHIFT+E |
| Vol Fade Out | CTRL+F | Thin Auto Events | CTRL+SHIFT+F |
| Vol Trim | CTRL+SHIFT+V | Context Sensitive | CTRL+2 |
| Trim | CTRL+T | Snap Point Edit | CTRL+SHIFT+D |
| Trim Adjacent | CTRL+ALT+T | Mix Edit | E |
| Scrub Solo | CTRL+B | Auto Enable | G |
| Scrub All | CTRL+SHIFT+B | ADC | CTRL+ALT+D |
| DC Offset | CTRL+SHIFT+N | Auto Snapshot | F |
| Normalize | CTRL+N | Zoom IN H | ALT+RIGHT |
| SRate | CTRL+SHIFT+S | Zoom OUT H | ALT+LEFT |
| Pitch | CTRL+SHIFT+P | Zoom IN V | ALT+UP |
| Time Stretch | CTRL+SHIFT+T | Zoom OUT V | ALT+DOWN |
| Reverse Tool | CTRL+SHIFT+R | Scroll Left | CTRL+ALT+LEFT |
| Mute | CTRL+U | Scroll Right | CTRL+ALT+RIGHT |
| Glue | CTRL+G | Scroll Up | CTRL+ALT+UP |
| Crop | CTRL+3 | Scroll Down | CTRL+ALT+DOWN |
| Solo | CTRL+H | Zoom Down | Z (1x=rect, 2x=prev) |
| Solo from Loc | CTRL+SHIFT+H | Zoom Up | SHIFT+Z (as above) |
| Consolidate | CTRL+SHIFT+G | Undo | CTRL+Z |
| Crossfade | CTRL+SHIFT+X | Redo | CTRL+Y |
| Mixdown | CTRL+SHIFT+Y | External Offline Process (Xpro) | CTRL+ALT+E |

| Various Mouse and Key Actions | |
|---|---|
| Create Record Track (Audio/Auto) | Right-Click RecTrkColumn (Left side Arr Window) |
| Centre Current Loc in Arr Window | C |
| Place Left Locator at Current Loc | L |
| Place Right Locator at Current Loc | R |
| Preview Take in File Manager | Right-Click+Hold on icon/name +move mouse |
| Add Marker at Current Loc | INSERT |
| Add Numbered Marker at Current Loc | (Enter Marker No)+INSERT |
| Move Current Loc to Marker | (Enter Marker No)+DOWN, or<br>Right-Click in Marker Window |
| Move Current Loc to Marker & Play | (Enter Marker No)+UP |
| Move Marker to Current Loc | (Enter Marker No)+ ENTER |
| Move Left Locator to here | Left-Click on Time Axis Bar |
| Move Right Locator to here | Right-Click on Time Axis Bar |

| Various Mouse and Key Actions (continued) | | |
|---|---|---|
| Move Current Loc to here | | SHIFT+Left-Click on Time Axis Bar |
| Move Current Loc to L Loc | | Left-Click on Left Loc Button (below Display) |
| Move Current Loc to R Loc | | Left-Click on Right Loc Button (below Display) |
| Song Position (User Enter Value) | | Left-Click on Song Position Time Display |
| Current Locator (User Enter Value) | | Left-Click on SMPTE Time Display |
| Left Locator (User Enter Value) | | Left-Click on Left Locator Time Display |
| Right Locator (User Enter Value) | | Left-Click on Right Loc Button (below Display) |
| Sign (User Enter Value) | | Left-Click on Sign Display |
| Tempo (User Enter Value) | | Left-Click on Tempo Display |
| Selct Multiple Parts in Arr | | Hold Down Ctrl, Left-Click on Parts |
| Selct all Parts from Sel Prt to End | | Hold Down Shift & Control, Left-Click on Part |
| DEL selct'd Parts (ArrWind active) | | DELETE key |
| DEL selct'd File/Folder (FileManWind active) | | DELETE key |
| DEL selct'd Marker(s) (MarkWin active) | | DELETE key |
| Mixer Inactive Build | | Hold SHIFT while editing mixer |
| (…to enable an Inactive Mixer) | | …last mixer edit w/o SHIFT<br>Double-Click Mix Window caption bar |
| Select tool to L or R mouse button | | Left-Click or Right-Click on tool in ToolBar |
| Select tool to Alt+L or Alt+R mouse | | ALT+Left-Click or ALT+Right-Click on tool in ToolBar |
| Up a tool page | | Right-Click on ToolBar page icon |
| Down a tool page | | Left-Click on ToolBar page icon |
| Customize Context Sensitive Edit Tool | | Double-Click on SwissArmyKnife Icon |
| Customize Cross-Fade Settings | | Double-Click on CrossFade Tool |
| Customize ToolBar Settings | | Double-Click on blank part of ToolBar |
| Zoom Box (for Arr… to zoom in) | | Left-Click Zoom Tool-> draw zoom |
| Zoom Out (for Arr… to zoom out) | | Double-Click ZoomTool |
| Magnify Waveform display in Part | | Right-Click on Zoom IN Vert |
| Un-Magnify Waveform display in Part | | Right-Click on Zoom OUT Vert |
| Select Multiple SOLO buttons in Mixer | | Hold CTRL while Left-Click or Right-Click solo buttons |
| Select default response to Pop Ups | | Spacebar or Enter |
| Cancel Processing Tools | | Spacebar or Enter |
| Nudging with Move or Copy on left Mouse | 1 Track Up | Left-Click+UP |
| | 1 Track Down | Left-Click+DOWN |
| | Previous Snap Point | Left-Click+LEFT |
| | Next Snap Point | Left-Click+RIGHT |
| Trim Tool on left Mouse | Trim Part Previous Snap Point | Left-Click+UP |
| | Trim Part to Next Snap Point | Left-Click+DOWN |
| Slip Tool on left Mouse | Slip Part Previous Snap Point | Left-Click+LEFT |
| | Slip Part Next Snap Point | Left-Click+RIGHT |
| Snap Point Edit Tool on left Mouse | Reset Snap Point | Double-Click on Part |
| Volume & Fade Tool; Noise Gate Parameters PopUp | Increase sel fader (fast) | Page Up |
| | Decrease sel fader (fast) | Page Down |
| | Increase sel fader (slow) | UP or LEFT |
| | Decrease sel fader (slow) | DOWN or RIGHT |
| Context Sensitive Edit Tool (SwissArmyKnife) | Floating Dropdown Menu | Right-Click on Part |
| | ZOOM | Right-Click Part, move mouse |
| | Scroll Part in Arr Window | ALT+Right-Click, move mouse |

# 7. Support

## Support, FAQs and Online Help Centre

To access the latest support information on SSL Soundscape and Mixer, MX4 or the SSL Console Bundle, please visit our online support site. The information there is kept up to date by our support staff to ensure it is accurate, and is available to you 24/7/365.

If you can't find an answer or a solution to your issue, you can submit a question via the site to our support staff for resolution:
www.solidstatelogic.com/support



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