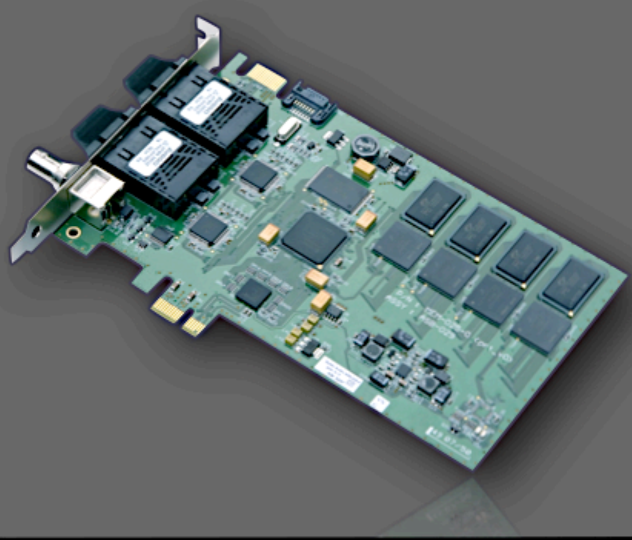
www.solid-state-logic.com

SOUNDSCAPE V6.0

The SSL DAW. For Windows PC's.

User and Reference Guide V6.01



SSL Soundscape V6. This is SSL.

Solid State Logic
SOUND || VISION

Table of Contents

1. Introduction

Welcome to SSL Soundscape V6, the High Performance DAW from SSL and the perfect companion for your SSL MX4 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 MX4, Solid State Logic's worldwide customer support team is easy to contact via the Support section of the SSL website and is always happy to help.

Please register your SSL audio card and the SSL Soundscape V6 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 SSL Soundscape V6 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.

However, SSL Soundscape V6 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 V6 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 V6, read the **"Soundscape V6 User Guide"** (Chapter4) as soon as you can.
If you are new to digital audio recording, reading the manual first is highly recommended.

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.

If possible: Please do not print this manual.

IMPORTANT: This Manual does **not** contain an **Index**. Please use your **PDF Reader's build in search function** to find the sections containing specific words and topics.

Reading conventions

Designation of supported hardware

The SSL Soundscape V6 software supports the MX4 and Mixpander audio cards. The functionality of these cards is similar although their specifications vary. The information in this manual relates to both cards. Differences are pointed out where necessary.

Legacy Soundscape Systems (eg. the Soundscape 32 System) can be used in a limited way with Soundscape V6 as a:

- 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 V6 Manual does not go into too much detail about installation and configuration of legacy Soundscape systems. Please read the Soundscape Editor V5.5 manual as well, when you plan to integrate a legacy SS system.

Key commands and key combinations

Some functions of the Soundscape V6 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 between square brackets. For example, the key for the letter "E" will be written: [E]. Key combinations will be written using "+" signs. For example, pressing the "D" key while holding the "Control" key will be written as [Ctrl]+[D].

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. This could be because your SSL hardware configuration is different, because you are using a different version of Windows, because you are using different Windows settings, or because the look of Soundscape V6 can vary (e.g., the Toolbar can be moved around the main window, there are different Skins to choose from).

Safety and Installation Considerations

General Safety

- Read these instructions.
- Keep these instructions.
- Heed all warnings.
- Follow all instructions.
- Do not use this apparatus near water.
- Do not expose this apparatus to rain or moisture.
- Do not block any ventilation openings. Install in accordance with the manufacturer's instructions.
- Do not install near any heat sources such as radiators, heat registers, stoves or other apparatus (including amplifiers) that produce heat.
- There are no user-adjustments, or user-serviceable items, on this apparatus.
- Adjustments or alterations to this apparatus may affect the performance such that safety and/or international compliance standards may no longer be met.

Caution

- To reduce the risk of electric shock, do not perform any servicing other than that contained in these Installation Instructions unless you are qualified to do so. Refer all servicing to qualified service personnel.

Installation Notes

- When installing this apparatus, place the host system into which it is to be installed on a secure level surface.
- To prevent damage from static electricity when installing this apparatus, either to the host system or to this apparatus, always take proper anti-static precautions. Always use an anti-static wristband. If in doubt, please refer to qualified service personnel.
- Take care of rough or sharp edges when accessing the inside of the host system.
- Never install or remove this apparatus whilst the host system is powered. Always remove the power cord from the host system prior to accessing this apparatus.
- If in doubt about installing this apparatus, please refer to qualified service personnel.

Disclaimer

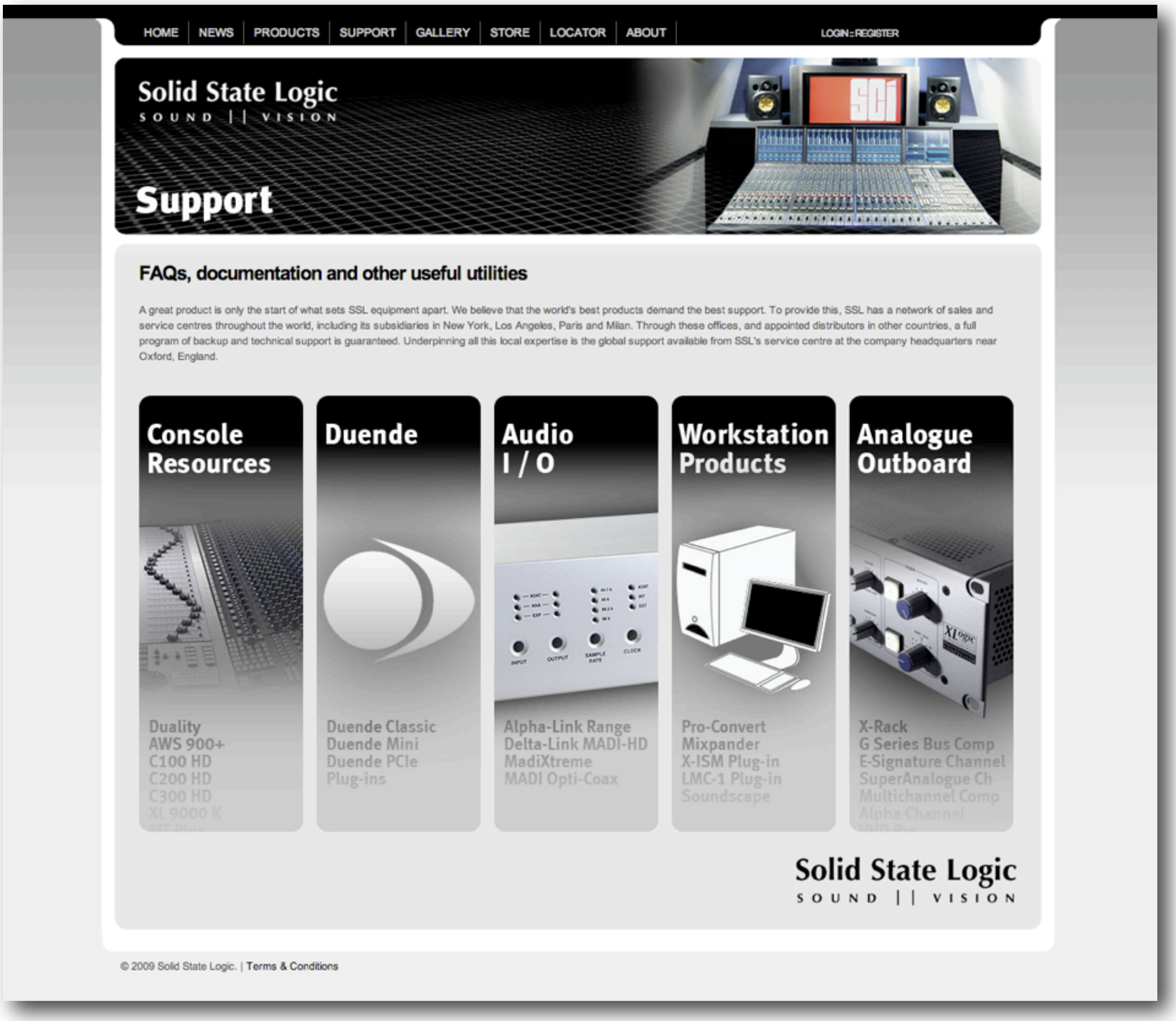
This manual has been written with great care and attention to detail, and we have attempted to cover every operational aspect of SOUNDCAPE V6. However, it is not a contractual document. Solid State Logic and/or the writer(s) of this manual cannot be held responsible for any loss or damage arising directly or indirectly from any error or omission in this manual.

Trademarks

All trademarks are the property of their respective owners and are hereby acknowledged.

Website

The URL for the Solid State Logic website is: http://www.solidstatelogic.com
The **SSL Support Website** is: http://solidstatelogic.com/support



2. SSL Soundscape V6 Software Installation

Important Preparations

Before you proceed installing **SSL Soundscape V6 software**, 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 SSL SOUNDSCAPE or Mixer V6 application.

The files required for the following Installation can either be found on the MX4 CD V2.0 (June 2010) or can be downloaded from the SSL website.
The MX4 CD comes with the **MX4 Install Menu Application** (MX4_start.exe), that 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 MX4 Hardware and Drivers properly. If you have an internet connection, please download and use the most recent version, available from the SSL website.
2. Run the SSL SOUNDSCAPE/MIXER V6 Combo Installer. (Both SSL Applications are installed with the same Installer)
3. If you want to remote control the SSL SOUNDSCAPE or Mixer V6 with a Hardware/Fader Remote, please also install the Console Manager Plug-In.
4. Start the SSL Soundscape V6 Software and follow the Instructions of the Unit Configurator. Make sure the Software runs properly.
5. Enter the Mix Password you find on your MX4 Registration card, under Options|Passwords|V2.xx Mixer starting with the letter M.
6. Enter the SSL Console EQ Password you find on your MX4 Registration card (SEQ), under Options|Passwords|SSL Console EQ starting with the letter M.
7. Enter the SSL Console Dynamics Password you find on your MX4 Registration card (SCD), under Options|Passwords|SSL Console Dynamics starting with the letter M.
8. Enter the SSL Bus Compressor Password you find on your MX4 Registration card (SBC), under Options|Passwords|SSL Console Bus Compressor starting with the letter M.
9. Enter the SSL Audio Toolbox Password you find on your MX4 Registration card (ATB), under Options|Passwords|Audio Toolbox (Partx) starting with the letter M. You only have to enter the password once.
10. Enter the SSL Soundscape V6 Password you find on your Soundscape V6 webshop receipt or in your MySSL account online (SS6), under Options|Passwords|Soundscape Version 6 starting with the letter M.
11. Enter the Time Module Password you find on your Soundscape V6 webshop receipt or in your MySSL account online (XTM), under Options|Passwords|XPro Time Module starting with the letter M.
12. Connect your Converter or MADI Console and set Clocking and MADI Mode under Settings|Master Clock and Settings|MADI|MADI Mode.
13. Ensure that the SSL Soundscape V6 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 V6 was designed to take advantage of the new SSL MX4 Card's DSP and Audio Core Technology and was optimised for modern Multi-CPU Processors and Windows 7.
For best results please 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.

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. Soundscape V6 can use up to two Mixpander Cards simultaneously.

NOTE: Soundscape V6 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 V6 setup, please visit our FAQ Section online, where you will find useful information and application notes. http://www.solidstatelogic.com/music/soundscape/faq.asp

System Requirements

The **SSL MX4 PCIe** card and the **Soundscape V6 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 SS V6
	Windows Vista SP1 or later	
	Windows 7 or later	
PC with 64 Bit Windows	Windows Vista SP1 or later	ASIO 32 Bit, SSL SS V6 (Support for ASIO64 and WDM 64 coming soon)
	Windows 7 or later	

MAC OSX: MX4 can run under OSX Tiger V10.4.11 or greater, on OSX Leopard V10.5.4 or greater and on OS X Snow Leopard V10.6.1 or later (32bit 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 however, do not work with MAC OS.

The **SSL MX4** 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.

The **SSL Mixpander PCI** card and **Soundscape V6 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 SS V6
	Windows Vista SP1 or later	
	Windows 7 or later	
PC with 64 Bit Windows	Windows Vista SP1 or later	ASIO 32 Bit, SSL SS V6 (Support for ASIO64 and WDM 64 coming soon)
	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.

System Recommendations

Soundscape V6 was designed for the **latest** PC Core Technology (CPU's, Chipsets, HDD's) and **Windows 7** Operating Systems.

As technology is constantly evolving, for up to date System Recommendations and a list of Soundscape V6 certified Turnkey Systems from 3rd Party vendors, please visit our System Spec page at: **http://www.solidstatelogic.com/music/soundscape/specs.asp**

Installing from CD or Web Download

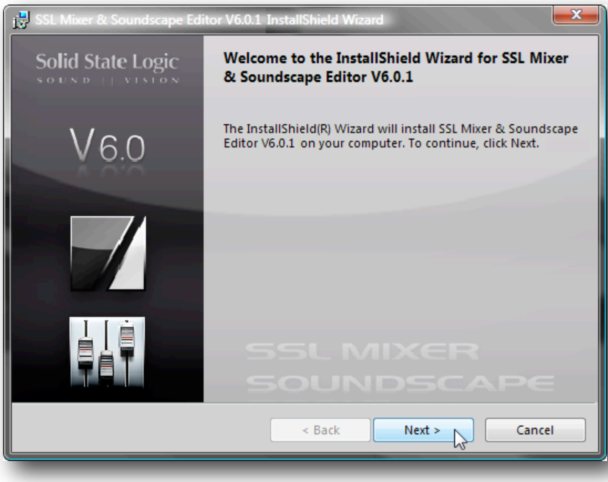
The MX4 Card comes with an Installation CD containing the SSL Mixer and Soundscape V6 Combo Installer.

Please insert the MX4 CD in your PC, let the MX4 Install Menu Application begin (Autostart for CD's needs to be active) and select **Install SSL Mixer or Soundscape V6**.

If your MX4 Installation CD has a Version prior to June 2010, or you are looking for a more recent Software Version, please **download the latest** Mixer/Soundscape V6 Combo Installer from our Website: **http://www.solidstatelogic.com/music/soundscape/downloads.asp**
Simply unpack the download into a folder and start Setup.exe

Once the Installer is started (depending on the Windows Version you may be prompted with some dialogue boxes to confirm that you really, really want to run this Installer)

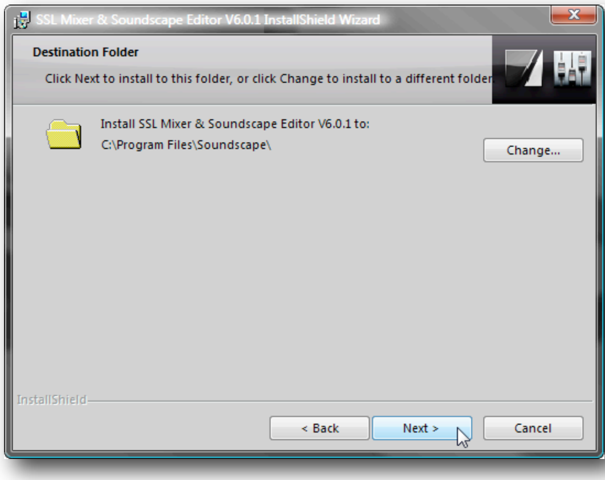
Click **Next.**



Please note that although you can select a different installation folder than the default offered, the SSL Mixer & Soundscape V6 user manuals will always refer to the default installation folders.
Also our Support docs and Support people will always refer to the default directories...

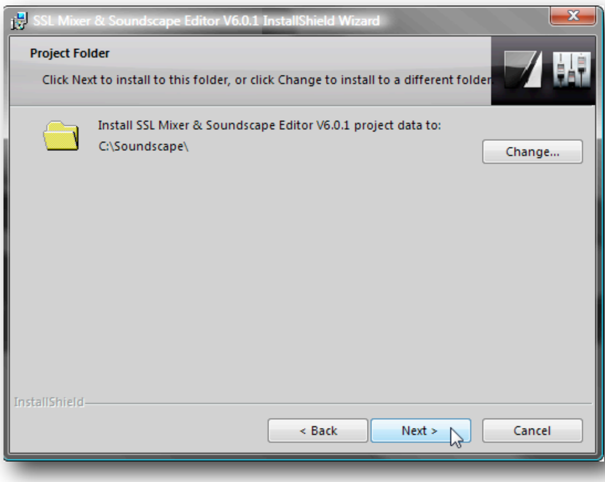
We therefore recommend using the default installation locations.

Click **Next.**

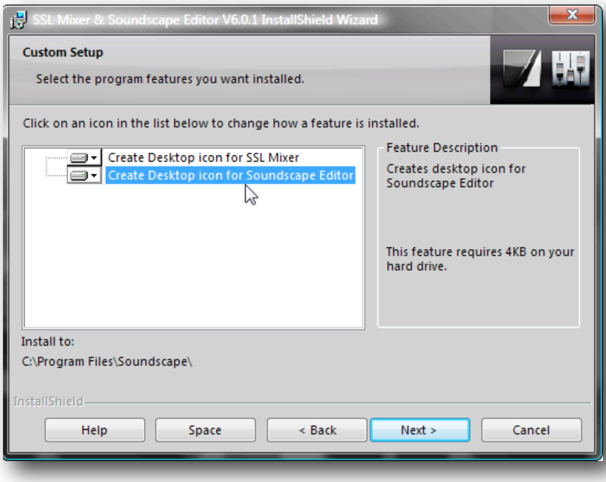


You can select a Projects Data Folder, we do recommend using the default location.

Click **Next.**

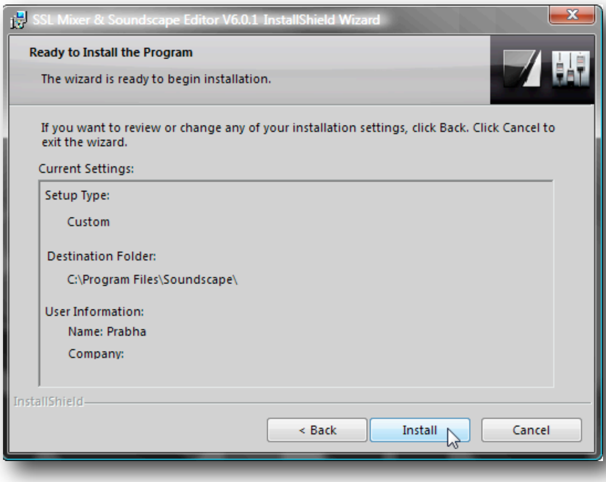


Once you have selected your installation and project data folders, you may select to install the SSL Mixer and/or the SSL Soundscape V6.



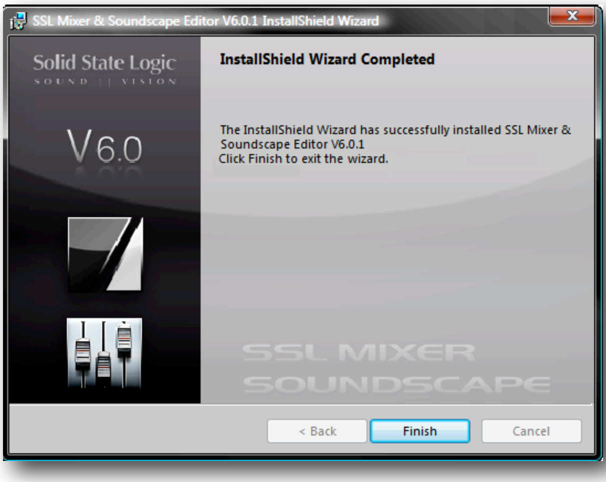
Click **Next.**

Now you are ready to install some very fine SSL Software...



Click **Install.**

Once the Installer has successfully finished, you can also....



Click **Finish.**

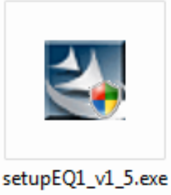
You are now ready to launch Soundscape V6 for the first time...

Happy SScaping...

Note: The SSL Mixer /Soundscape V6 Combo Installer will also install the latest Version of the **SSL Console Bundle** and **Audio Toolbox.**

Installing optional DSP and VST Plug-Ins

SSL Soundscape format DSP plug-ins



Installing SSL Soundscape plug-ins is a straightforward process.
First close all the SSL Soundscape or Mixer Applications, double click on the plug-in's installer application icon and follow the instructions.

If you have installed SSL Soundscape V6 in its default folder, you will not need to interact at all, if Soundscape V6 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 V6 and enter the Plug-In Password you have received by email into the appropriate Line in the Options Menu.

Authorising Plug-ins in the Options Menu is explained in Detail in **Chapter 5 › Options Menu.**

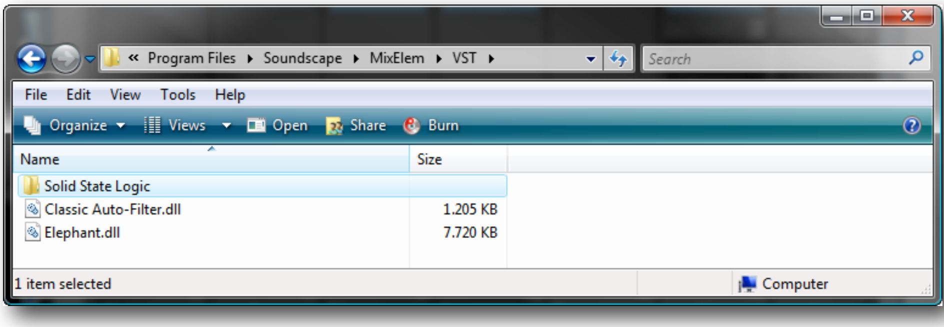
Please note that when working with Windows Vista or Windows 7, you may have to run the installer in compatibility mode for Windows XP (right-click Installer Program->Properties->Compatibility Tab).

VST format plug-ins

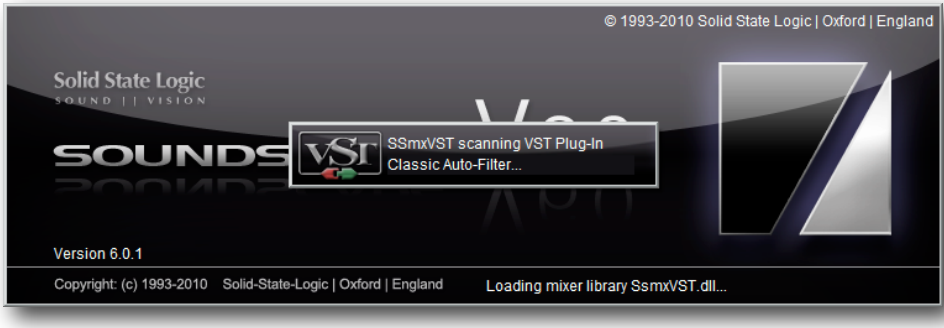
The process of installing VST format plug-ins may vary depending on the developer's installation process.
However, many VST Manufacturers install the VST.DLL files by default in: C:\Program Files\Steinberg\VstPlugins.

For Soundscape V6 to be able to work with installed VST plug-ins, their DLL files must be present in SSL Soundscape V6's default VST folder, which is under normal circumstances (=you used our default locations to Install Soundscape V6) :
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.



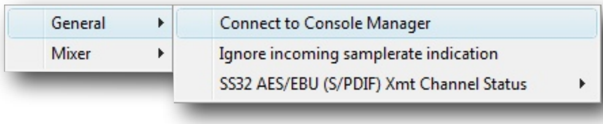
At startup, Soundscape scans the VST folder and subfolders (Solid State Logic in the example above) and a small window appears in front of the splash screen where the plug-in name is displayed while loaded.



Installing the SSL Console Manager

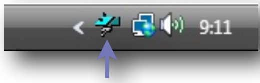
Double Click the **Setup Console Manager 1.55** icon on the MX4 Install Menu (or the Setup.exe inside the folder you extracted the CM155.zip web download to) to install the Console Manager software on your computer.
Please follow the on screen instructions of the Installation wizard.

Enabling Console Manager in SSL Mixer V6
Console Manager can not be launched as a standalone program.
It is a software module of the SSL Mixer V6.
Enabling the entry **Connect to Console Manager** in the Menu **Settings|Preferences|General** opens the module automatically and connects it to the Mixer Software.



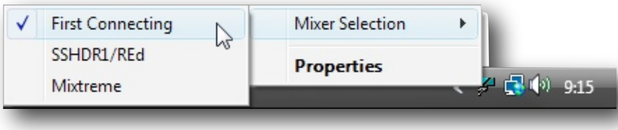
NOTE: If you always want to invoke the Console Manager Module when you start up the SSL Mixer V6, please activate **Connect to Console Manager** followed by a **Save Settings** command in the Menu **Settings|Save Settings**.

Console Manager Taskbar Icon
An icon appears in the taskbar once the Console Manager is running.



Console Manager

Right-clicking on the task bar icon opens the following menu:



Mixer selection opens a submenu to chose to which software the Console Manager should be connected to:

- **First Connecting:**
  The first SSL/Soundscape Software that invokes Console Manager gets exclusively connected (recommended setting)
- **SSHDR1/REd:**
  Use this entry to connect legacy Soundscape Editor Software to the Console Manager Module
- **Mixtreme:**
  Use this entry to connect legacy Soundscape Mixtreme and Soundscape Mixpander powerpacks under SS Mixer 4.3 or below to Console Manager

To use Console Manager with the SSL Mixer V6, **First Connecting** should be activated.

NOTE: The Hardware Controller you want to use with Console Manager must be connected and switched on before the Console Manager program is launched. The program may not detect the device if it is switched on afterwards.

High Performance SSL Audio Engine

SSL Soundscape V6 is wholly new DAW Software 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 V6 is made for speed and to keep the user in a creative flow.

SSL Soundscape V6 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 V6 allows safe collaboration over a standard Gigabit Ethernet Network.

Digital mixing, effects and processing

The SSL "Console" inside the Soundscape software runs on the on-board DSP-powered mixing engine providing immense audio processing capabilities. The Mixer's architecture is amazingly flexible and puts no limits on the way 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, also known as "latency". This may be very small on an expertly configured, modern 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, by its very nature, 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 (i.e. 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 V6 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 (e.g., during mixing and mastering) and allows access to hundreds (if not thousands) of plug-ins.

3. First Contact - the V6 Quick Start Guide

The purpose of this chapter is to quickly get you going with SSL's Soundscape V6 DAW.

You will learn how to initially configure your system, create our first project and we will introduce you to a handful of Soundscape V6 specific terms and workflows.

Experienced Soundscape Editor Operators will learn at the end of this Chapter how to **Migrate their Projects to V6.**

This chapter does not explain functionality in Detail.

In order to explore Soundscape V6 and its wealth of useful features, please read **Chapter 4** for a more detailed overview and then **Chapter 5** to really understand the unique Soundscape V6 concepts and how they help you to speed up your audio life function by function.

Curious? Let's dive right in....

SSL Mixer or SSL Soundscape V6?

The **SSL Mixer/Soundscape V6 Combo Setup** installs one or two applications (depending on your choice during setup).
Both applications 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
Its main purpose is to work with other audio software like Cubase/Nuendo, Sonar, Reason, etc. and take care of Routing and processing audio before or after the DAW's software input and perform mixing and monitoring tasks in "DSP Realtime" (4 samples).
It also may 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 V6
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 V6, the Mixer is already built inside!**
The seamless collaboration between DAW and Mixing functionality inside Soundscape V6 is ideal for mission critical applications including Live Recording, high channel count Music Production, Broadcast, Post and Restoration.

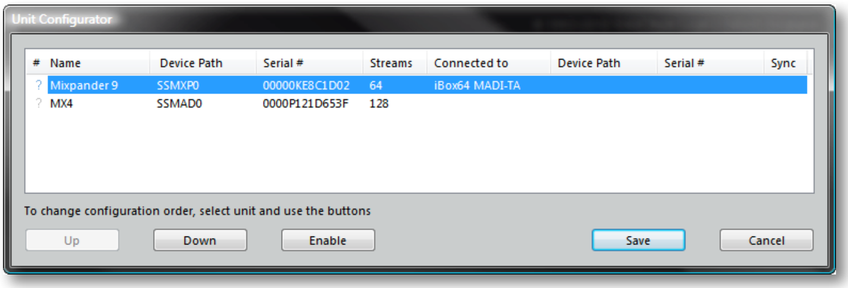
SSL Soundscape V6 and SSL Mixer together
They may run simultaneously if necessary, mainly when two different and independent setups are desired. In this way you may use the SSL Mixer to perform sub mixing tasks from live inputs or sequencing software and at the same time run the SSL Soundscape to play backing tracks, record live or edit in realtime.
It is important to mention that each software entity **is using any SSL Card exclusively**, so you need at least two SSL cards to run Mixer and SOUNDSCAPE simultaneously (i.e. a MX4 and a Mixpander, both connected to an Alpha Link or MADI iBox interface).

To **assign each software to work on a different card**, you must configure the Unit Configurator utility under the Settings menu of the SSL Mixer and SSL Soundscape as explained below.

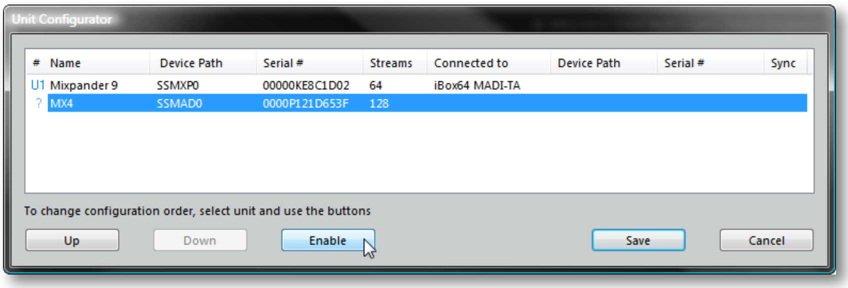
Unit Configurator

When SSL Soundscape V6 runs for the first time it will automatically open the Unit Configurator window to select and configure the Hardware you want to use.

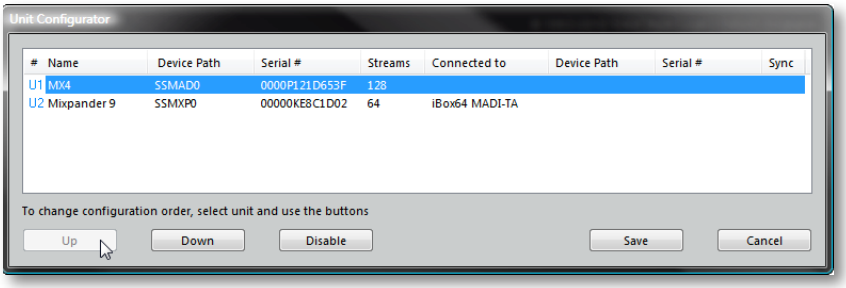
Initially all SSL units in the system are disabled.



To activate a unit highlight it and click the button Enable

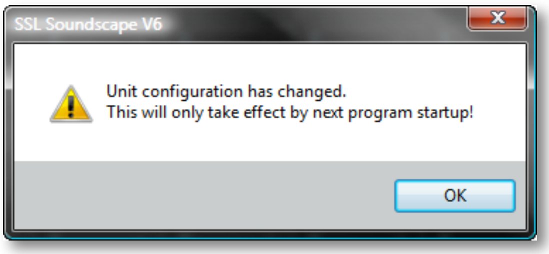


With the Up and Down Buttons active Units can be renumbered, so they are seen by Soundscape V6 according to your preferences as unit 1, unit 2 and so on. The Top Unit is always U1, the second Unit always U2 a.s.o.



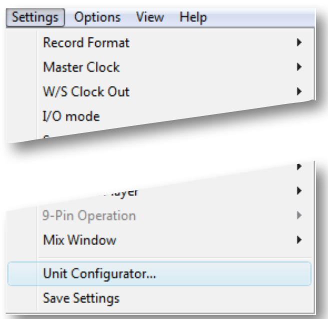
When you are happy with the unit configuration, press **Save**.

A warning message will appear:
In order for your configuration to become active, the SSL Soundscape V6 software must be restarted.



After re-start, SSL Soundscape V6 will launch into the Main Window.

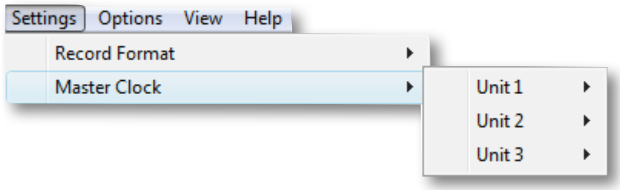
Opening Unit Configurator again
If you made an error or your configuration is not working as expected, you can launch the Unit Configurator again from the Settings menu.



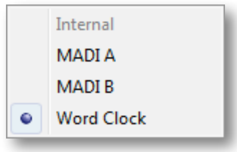
Clock Synchronisation

Any digital audio System must be properly clocked to a common clock source.
In an MX4/Single Alpha Link Configuration either the MX4 needs to be locked to the Alpha Link or vice versa.
When both MX4 and Alpha Link are clocked internally (=no Synchronisation of the clocks) you will hear audible clicks whenever the MADI Frames get too far off.

The SSL Soundscape V6 Master Clock Setting changes the Clock Source for the MX4:

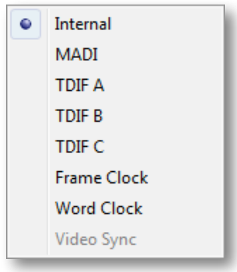


In a System with a single **MX4** the choices on the right are available. The MX4 becomes clock master by selecting "Internal". Alternatively it can become a clock slave to the connected audio I/O by choosing "MADI A, MADI B or Word Clock".



A connected Alpha-Link needs to be clocked appropriately. If the MX4 is set to Word-Clock, a BNC Cable from the Alpha-Links Word-Clock Out needs to be connected to the MX4's Word-clock Connector.

A **single Mixpander** system becomes clock master by selecting "Internal". Or set it to be clock slaved to the connected Alpha-Link simply by 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 a particular system.
The Word-Clock connectors are an obvious solution. 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 SSL Soundscape and Mixer V6 software can work with any combination of MX4 and Mixpander cards. Within the Mixer environment the cards are identified as Unit 1, Unit 2, Unit 3 ... .
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 Alpha Link to Internal Clock.

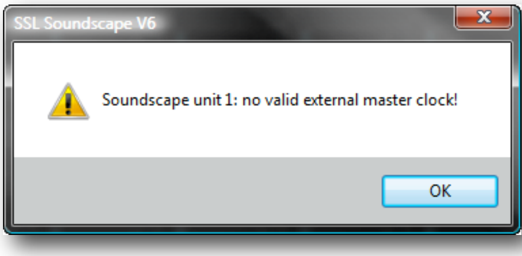
For a multiple MX4/Mixpander system, units 2 and above must be operated as a clock slaves, receiving the Clock signal from another unit via their Word Clock or MADI input. The unit that provides the Clock signal must be set to output "Worm Clock". Worm Clock is a modified Word Clock signal that includes 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 "Settings Menu" 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 following dialog box appears, check your digital out to digital in connections, as the unit indicated will not be receiving a valid clock signal at its selected digital input. Please also check the “Master Clock” and “W/S Clock Out” sections mentioned in **Chapter 5 › Settings Menu.**

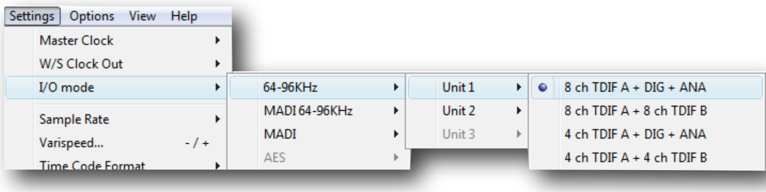


IMPORTANT: Legacy Soundscape hardware must always be configured as **Unit 1.**

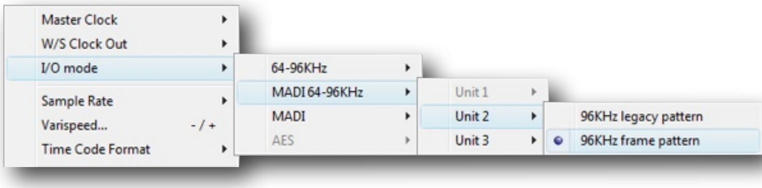
I/O Mode

The I/O Mode Entry in the Settings Menu provides several submenus to setup hardware specific operation at high Sample Rates and MADI Ch Modes.
The available options are determined by the SSL Hardware you are using.

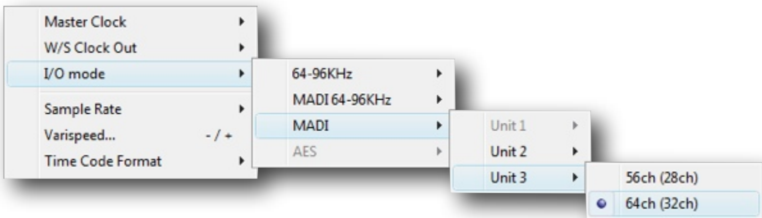
64-96kHz
Allows selection of different **I/O configurations** and **interleaving** options at **high sample rates,** such as SMUXed or non-SMUXed Modes for ADAT/TDIF/MADI/AES or different **I/O combinations** for a Soundscape 32/Mixpander system.



MADI 64-96kHz
In this submenu the MADI Frame Pattern is selected between **96k Legacy Pattern** 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. In the beginning **MADI had 56 CH** (at 44.1kHz and 48kHz) leaving some headroom in the data bandwidth to "Vari-Speed" all Channels at a higher play speed (= higher Sample Rate).



Later the **64 CH MADI Mode** was introduced as an AES standard, which always uses the full MADI data bandwidth, hence doesn‘t allow a higher sample rate.
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 CH MADI** cable operates with **28Ch** of audio, while at **64 CH MADI 32 Channels** of audio are transmitted.

SOUNDS STRANGE? By default the SSL Alpha Link‘s MADI is set to 56Ch MADI Mode. If you experience a "comb-filter“ like sound in combination with an MX4 Card, please change the MX4‘s MADI I/O to 56 CH or consult the Alpha-Links Manual on how to change its MADI I/O to 64Ch MADI.

Time-Code Synchronisation

Since MX4 and Mixpander Cards do not have Time Code synchronisation facilities, external Time Code synchronisation is only available when a Soundscape 32 unit connected to a PCI Host Card is present in the system as U1.
However, the ASIO Positioning Protocol may be used with an MX4/Mixpander based system configuration, allowing APP compatible applications (i.e., Steinberg Cubase or Nuendo) to be slave-synced to Soundscape V6 sample-accurately.

For an SSL Soundscape system with a Soundscape 32 and Mixpander card the synchronisation information is passed to units 2 and above by setting each unit's W/S Clock Out to Wormclock.

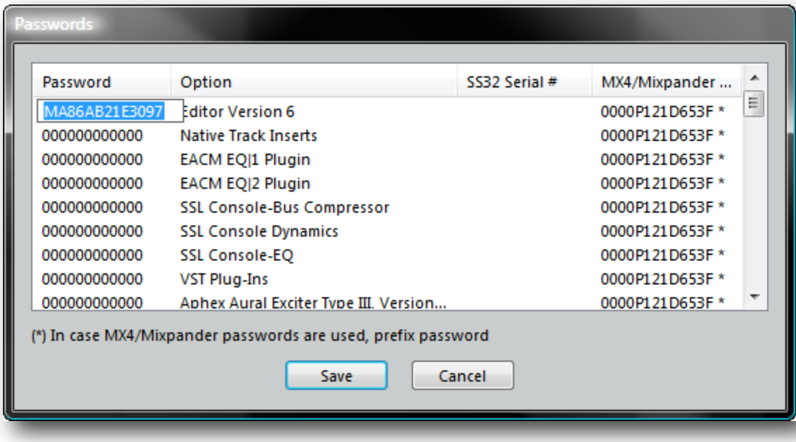
For detailed information about the Sync In options, please read the Synchronisation section of the Menu Reference chapter.

Authorising the Soundscape V6 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), and any other plug-ins like 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 unit at our website, they are also available in your MySSL account:
http://store.solidstatelogic.com/user

If you have purchased any optional effects plug-ins, this is also the place where you enter their passwords. Please check the **Options Menu** section of the Menu Reference (Chapter 5) if you need more help.



IMPORTANT: For MX4 and Mixpander all passwords need to be entered with the prefix “M”.

If you haven't entered a correct password for:

- **V2. Mixer (MIX)**: You will not hear any sound or be able to record or play back. Soundscape V6 will notify you with an "Invalid Optional Module Password Message" and deactivate the Mixer (Mixer Inactive).
- **Soundscape V6 (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 V6.
  However, if you haven't purchased Soundscape V6 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 MX4 included Plug-Ins that are automatically installed with Soundscape V6. If you load the default or any Standard Mixer or insert one of those Plug-Ins into a Channelstrip, Soundscape V6 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.

Open an existing Mixer and Arrangement

SSL Soundscape V6 uses separate files for the Mixer and for the Arrangement, allowing for maximum flexibility.
To get an idea of how SSL Soundscape V6 works, you may open pre-made Mixer and Arrangement files.

You can download sample Soundscape V6 Projects here:
http://www.solidstatelogic.com/music/soundscape/downloads.asp

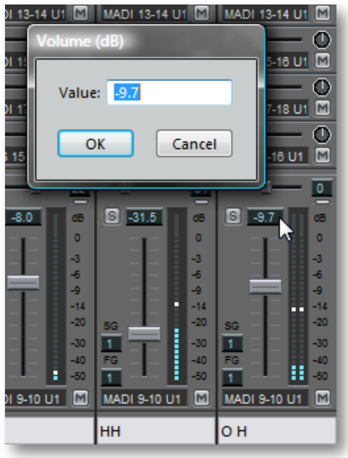
Click "Open" under the File menu, locate and load the Mixer file.
Click "Open" under the File menu, locate and load the Arrangement file.

Use your mouse to move around the screen and get familiar with selecting tools and pulling down menus.

To perform some edits on the audio takes, click on any icon located in the Toolbar with either mouse button to assign it to that 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 to select two more active tools, which will be 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 by grabbing them with the mouse, by positioning the mouse over the controls or value displays and moving the scroll wheel, and by double clicking on them and entering the desired value.

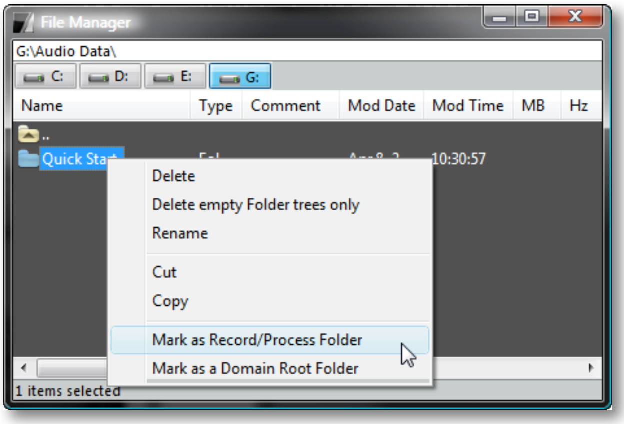


NOTE: Several Mixer files are installed with SSL Soundscape V6. If you installed using the default settings the Mixer files' path is "C:\Soundscape\Mix\" and using "Open Mix" under the File menu will automatically find this folder.

Recording Audio

Your Red Light is On!

1 Open Soundscape V6 and navigate to the File Manager on the right. (if you don't see the File Manager press [D])

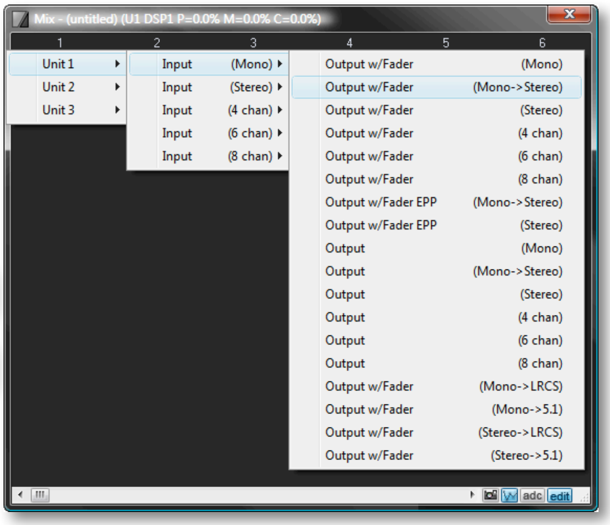


Navigate to the Drive you want to record to and create a new folder in the File Manager by right clicking in the empty space of the List View and selecting **Create New Folder** (why not call it Quick Start;-).

Right click on this Folder and select Mark as **Record/Process Folder** so Soundscape V6 knows where to store recorded Takes.

2 Create a new Arrangement by selecting the **"New Arr..."** option under the File menu.

3 Select **"New Mix..."** in the the File menu. You will be presented with a blank Mixer window:



4 Enter **"Mixer Edit" mode by pressing [E]** on your keyboard or by clicking on the "edit" button (right bottom of the Mixer window)

5 **Left-click** on the **Create tool** (left mouse button):

...and **right-click** on the **Track Assign tool** icon:

This way the two tools you need 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).
Create a **"Input (Stereo)" -> "Output with 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)"**

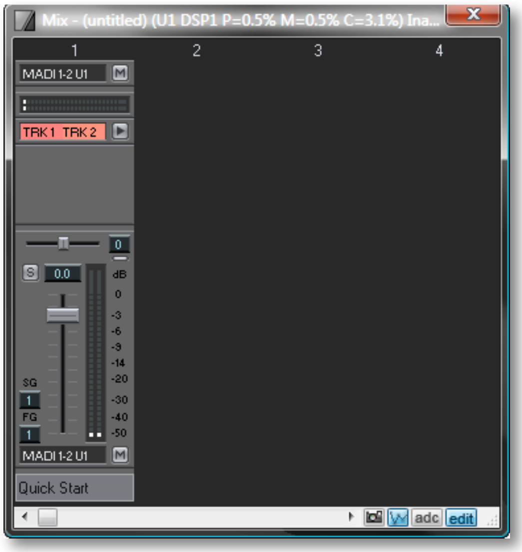
8 **Left-click below the peakmeter** to create a native stereo track insert.

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.

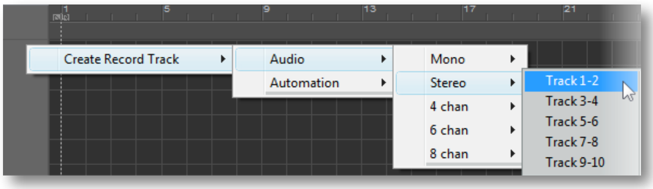
Press [E] on the computer keyboard to toggle between "Mixer Edit" mode and "Mixer Control" mode (or click on the Mix Control/Edit toggle instead).
Now double-click into the field at the bottom of the mixer column to open a dialog box to enter a name for the column. For this tutorial we have entered the name "Quick Start".

The Mixer should look similar to this in small view mode (press [Q] to show it or hide it):

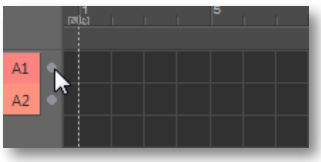
...and similar to this in full view mode (press [X] to show it or hide it):

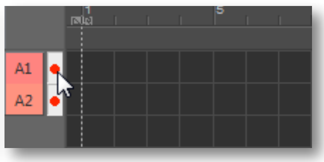


10 **Right-click anywhere in the Record Track Column** at the left side of the Arrange window. A menu will appear. Select "Audio", "Stereo" and "Track 1-2" .



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" (clicking on any one of the dots will activate both record tracks because this is a stereo pair).



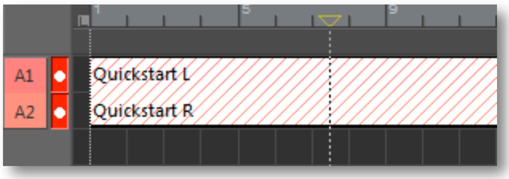


These buttons are linked to the button on the track insert element of mixer column 1.

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 lit, otherwise you could get some very nasty digital distortion.
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 input 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 "Settings" Section in the Menu Reference (Chapter 5) for details.

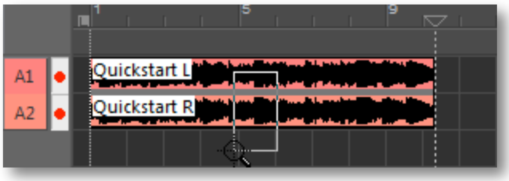
SSL Soundscape V6 will create two temporary Parts (shaded), and will record from the current play position until you press Stop on the Tape Transport (or the [Down Arrow] key on your computer keyboard). When recording starts, the Time Axis will appear shaded in the recording range and the track arming 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.



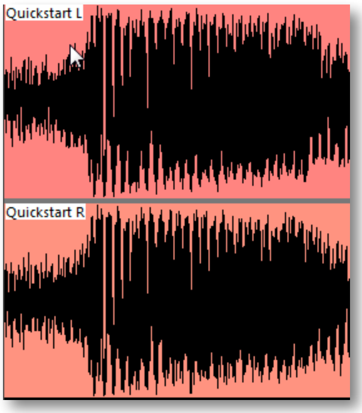
When recording is stopped, the recorded areas, known as Parts, will turn to solid colours and waveforms will be generated. You can move, copy, cut, trim, fade, etc., a Part without altering the original Take stored on the Disk.

Let's have a look at the audio you have just recorded at waveform level.

12 Hit the **[Z] key** on your computer keyboard and the mouse pointer will change into a magnifying glass. By drawing the outline of a selection box you indicate the area you want to zoom in on.



Try and zoom into a small area of the newly recorded Part. You should see a waveform similar to this.



NOTE: Hitting the [Z] key twice will zoom and position to the previous View (Soundscape V6 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 the [Z] twice to get to the previous view).

14 Select the **Copy tool**

...and "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 set to "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.

14 To name the Part, select the **Info tool**:

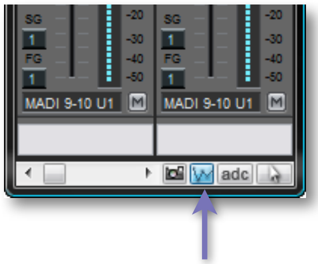
...and click the Part you want to rename. As soon as the info box appears you can type in a new name. There are many other parameters you can alter from within the Part Info window, as described in the "Editing Tools" chapter. After typing the new Part name, click "OK" to return to the Arrange window.

Recording Automation

Recording Automation data is very similar to audio recording and it shares almost the same steps. 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, plus three specific tools which are described in the **Mixer Automation section** of Chapter 4: SSL Soundscape V6 User Guide.
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 Enabling Automation

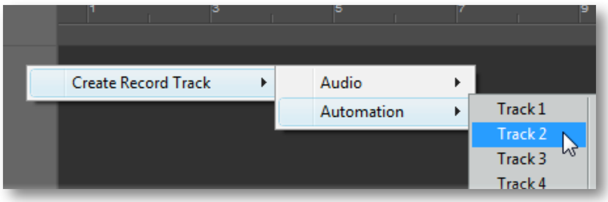


For any automation data to be recorded or played back, Automation must be enabled. This can be done by clicking the Automation Enable toggle located in the bottom, right corner of the Mixer window or you may press [G] on the computer keyboard. When the button is "illuminated" in blue, Automation is globally enabled.

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

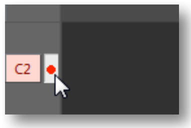
2 Create an automation Record Track

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



3 Arm the Automation Track

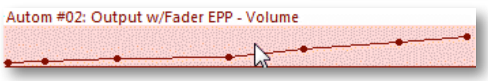
The track arming button works just like an Audio Track Arming button.



4 Record Automation

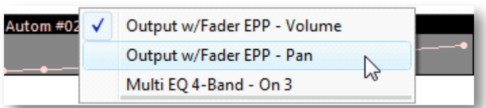
Make sure no audio tracks are armed, start the automation recording by pressing the record button or [+] on your keyboard and move the desired knobs on the Column No. you are currently recording. You can also create multiple automation tracks to record dynamic changes on multiple Mixer Columns.

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

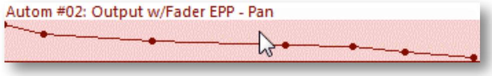


6 Curve Select Tool

With the Automation Curve Select tool you can select the automation curve to be displayed for each knob that has been moved during the automation recording.

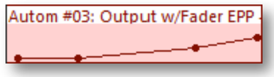


7 Automation Events Thinning Tool



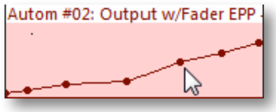
After clicking on an automation part, you may enter a thinning factor to reduce the number of automation nodes/points and make it easier to edit the curves graphically. A thinning factor of 200 reduces the amount of automation nodes slightly, a factor of 1000 reduces the amount of nodes heavily and only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

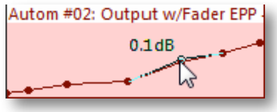
In this example the left automation part was recorded using a motorised Fader on an SSL Console to perform an S-Curve like Fade In. After applying a Thin Factor of 1000 only the shallow slope at the beginning, the steeper slope in the middle and the shallower slope close before the Maximum are maintained in the right automation part.



8 Automation Event Editing tool

The automation curves may be edited by clicking and dragging the automation nodes.





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 (i.e., 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".

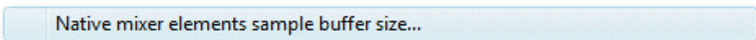
Mixing, VST Buffers and ADC

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 from the rest, causing phase problems and even an audible misalignment, if the processing takes a long time.

While all Soundscape V6 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 process" (either a rack mount FX Device connected to the Converter by "audio cables" or a "native" VST/VSTi Plug-In inserted into the Mixer) will add processing and/or converter latencies.

By inserting VST processes this delay is most noticeable, since the audio has to travel to the computer’s CPU, the CPU needs to buffer some audio for processing (it also has to perform many other tasks at the same time) and back to the Soundscape DSP's.

The amount of buffering for the CPU is user definable with the “Native mixer elements sample buffer size” option value in the SettingsI Preferences| Mixer menu.

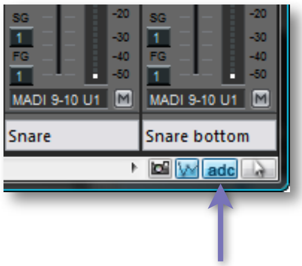


The default value is 128 samples, which should provide a comfortable time for a modern CPU to process any modern Plug-In. If you hear audible clicks or crackles or feel that the PC is struggling to cope with the processing demands, increase this value in powers of two up to 8192 samples. (ie. 64, 128,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

If an invalid value is entered, the closest valid value will be used instead.
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 a 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 per Plug-In, resulting in multiple processing delays.



Large Processing Delays can be solved by adding compensating delays to some of the "faster" signal paths in order to get the audio signals perfectly aligned at the outputs. To activate the **Automatic Delay Compensation** on the SSL Soundscape V6 Mixer, click the “adc” button located in the right bottom corner of the Mixer window. When the ADC Button is illuminated in blue, the automatic delay 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 (i.e., the one with the highest cumulated processing delay). Any signal going through the system may then be delayed including the artist monitoring paths. Soundscape V6 offers a variety of strategies to exclude certain audio paths (like Monitoring) from being delayed by the ADC. Please read the Section "Automatic Delay Compensation" in **Chapter4 › SSL Soundscape V6 User Guide** for more information.

Migrating Arrangements and Mixes from previous Soundscape Editor Versions

With SSL Soundscape V6 the proprietary SDisk format, known from any previous Soundscape Editor Version, is not recognised anymore.
SSL Soundscape V6 now accepts any logical drive recognised by the Windows operating system, which was not possible with any previous Soundscape Editor Version.
This move is a beneficial development in many areas, like 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 as well.

In order to prepare all SDisk Contents to be used with SSL Soundscape V6, the Takes, Arrangements and Mixers need to be copied to standard Windows Drives (FAT16/32/NTFS/NFS etc.).

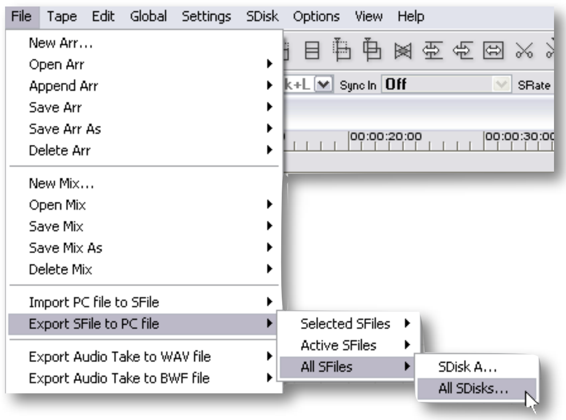
Please install the Soundscape Editor V5.5, for the file transfer process you do not need any Soundscape Legacy Hardware.
Please follow the instructions in the **Soundscape Editor V5.5 Manual** on how to setup and connect an **SDisk as a Native Unit.**

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 residing on 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.

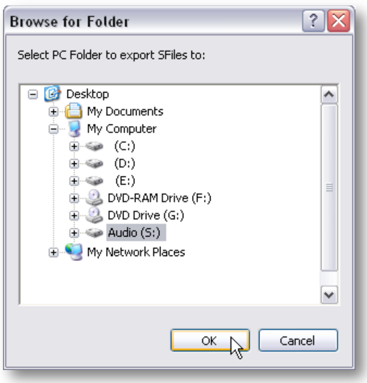
1. Open the legacy Soundscape Editor



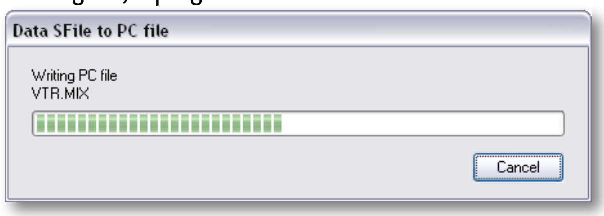
2. Go to the File menu and select
   Export SFile to PC File › All SFiles › All SDisks

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 File Copy to your Windows Drives is done, you can close the legacy Soundscape Editor and open SSL Soundscape V6, which can now open all your arrangements and mixer files. (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: **http://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 V6 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 V6 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 V6 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 V6 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 V6 Arrangement Format) to avoid any Search times next time you need to open it.

- If you plan to share files between SSL Soundscape V6 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 SSL Soundscape Editor and also the SSL Mixer from V2.x and above.

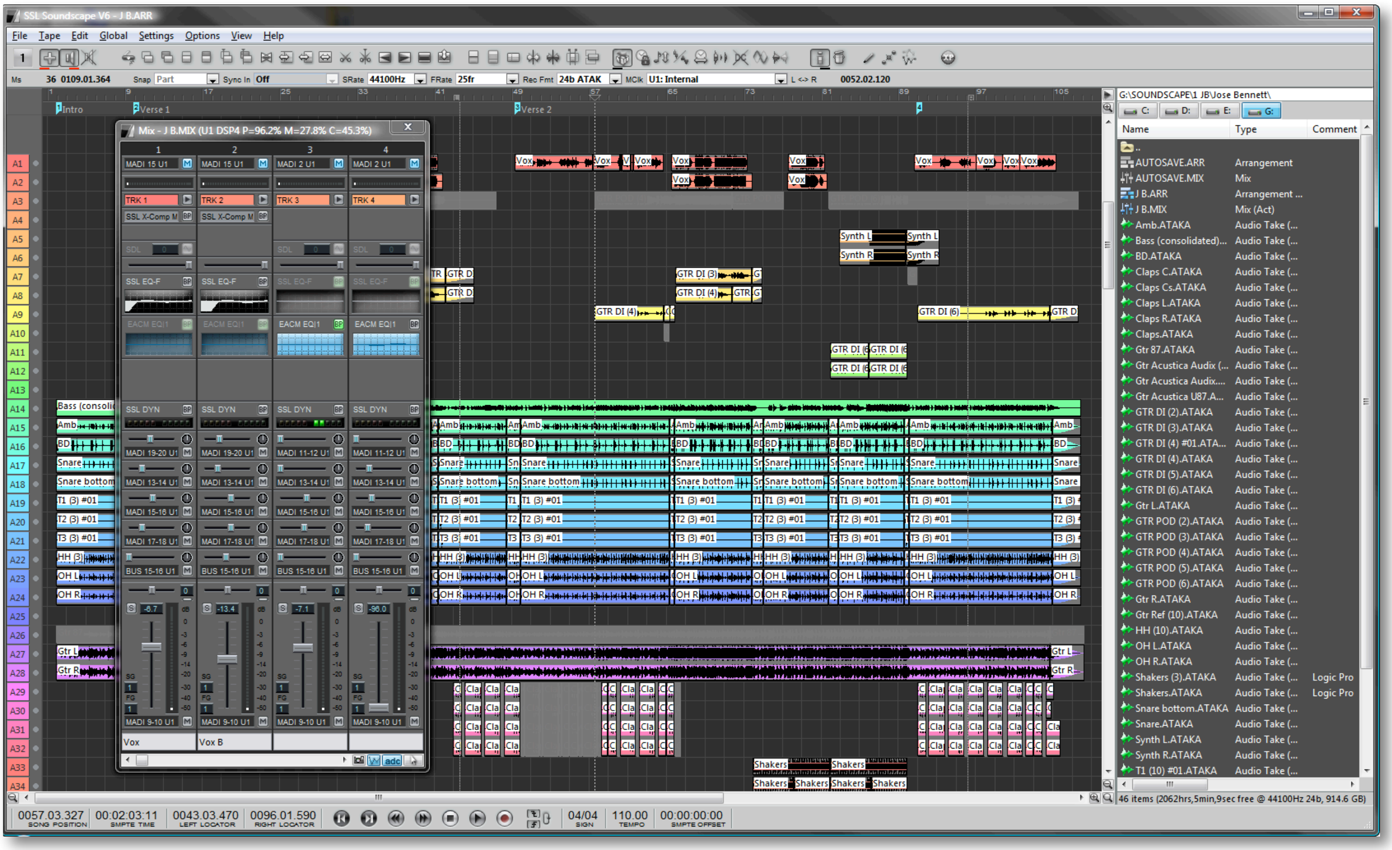
IMPORTANT: In Soundscape V6 the SDisk Menu is gone. In order to assign the Record/Process folders for an arrangement when it's opened for the first time, right click on your desired folder and select Mark as Record/Process Folder from the menu.
This Folder assignment is stored inside the Arrangement File.

Main Screen Overview

The main screen is customisable and you can hose to only display the Soundscape function Windows you find useful, like:

Arrange Window, Mixer Window, File Manager Window, Marker Directory, Video File Player, Big Current Time, Buffer Activity, 9-Pin Sync, and the main Soundscape V6 Window with 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.

Arrange and File Manager Windows can be docked to the Main Window.

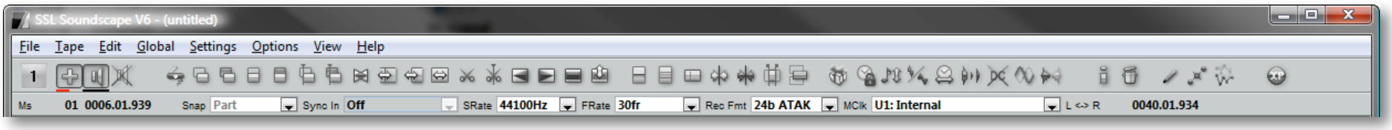


For Multi Monitor Setups all Soundscape V6 windows can be freely placed anywhere inside the Extended Desktop.
A Soundscape V6 Timeline Project is called Arrangement, has 256 virtual Tracks to create adventurous projects and can contain up to 16384 individual Parts (Blocks of Audio on the timeline) which use up to 16384 audio Takes (Audio Files on the Disk). An Arrangement can be created from scratch, or a saved Arrange file (.ARR) can be loaded.

The extremely flexible Mixer in Soundscape V6 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.
And they **should be backed up:**

IMPORTANT: Please do BACKUP frequently! (we remind you again later!)
If you don't backup... you risk losing your valuable work. Drive Recovery is extremely expensive!

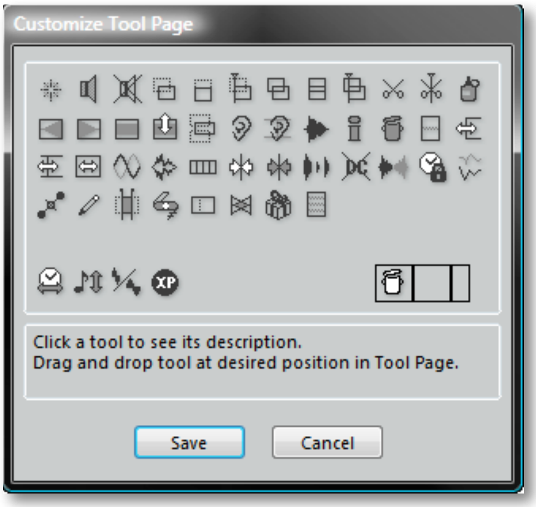
Toolbar



The top of the main Soundscape window contains the main menus, the Toolbar that is used for most on-screen editing, and the Status Bar.
The main menus are explained in detail in **Chapter 5 Menu Reference.**

Toolbar ([Page Up]/[Page Down] to scroll through the different Tool Pages)
There are 9 customisable Tool Pages, any of which can be displayed in the Toolbar. Scrolling through the Tool Pages can be done by left or right-clicking the page number icon, or by pressing the [Page Up] and [Page Down] keys on the computer keyboard.

NOTE: It is possible to go directly from page 1 to page 9, or from page 9 back to page 1, without having to scroll 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, which can be used for customising the currently displayed Tool Page.

Clicking on any tool icon will display a description of its function. Tool icons can be dragged and dropped at the desired position in the Toolbar and existing tools will be shifted laterally to make room for a new tool if necessary.
To delete a tool from the displayed Tool Page, drag and drop the white trash can icon over it. The other tools will then be shifted to the left to close up the space.
To insert a space between tools, drag and drop the full space or half space icons (empty boxes beside the trash can icon) at the required position.
Once you are happy with the Tool Page, click the "Save" button. This will save all changes you have made to the current Tool Page globally.

Tool selection

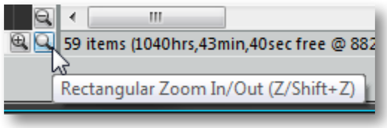
Four tools can be assigned to the mouse buttons, as shown above. Click on the tool of your choice with the mouse button you want to assign it to. A black bar will be displayed below the selected tool's icon, to the left or right side according to the chosen mouse button. You can select two other tools 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 the corresponding mouse button.

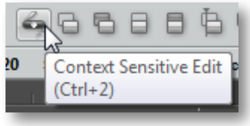
NOTES:
- A tool will remain selected even when a different Tool Page is displayed, that does not contain this Tool Button.
- If the same tool is assigned to a mouse button and the "[Alt] key + same mouse button", only the black bar will be shown.
- Selecting the "Context Sensitive Edit" tool will automatically cancel all previous tool selections. This is normal behaviour since this Tool uses all 4 Mouse functions.

Tooltips

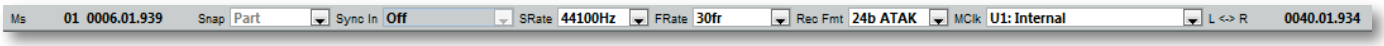
When the mouse pointer is hovering over any editing tool or any other function button, a tooltip is displayed after approx. 1 sec.

Where applicable, the corresponding key command is shown.





Status Bar



The Status Bar, displayed underneath the Toolbar, provides information and allows quick selection for some of the more commonly accessed parameters from the Settings menu. It 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.

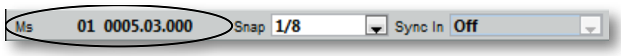
NOTE: The Status Bar can be turned on or off in the View menu.

Mouse position

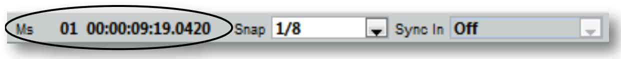
The mouse position readout is continuously updated to the current position of the mouse pointer in the Arrange window. Its resolution is determined by the snap setting (if active), and the Time Axis setting (SMPTE or bars/counts/ticks) determines the units the time position is displayed in. In SMPTE mode the sample within each frame is also shown.
The number displayed on the far left indicates which “virtual track” (horizontal position) the mouse is currently placed 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 (e.g. 1650 or 2560 pixels), so do not expect to be able to move in steps of 1 frame when you are displaying 5 minutes of the Arrangement on screen. However if you “zoom in” the mouse position readout will be as expected.

Bars/counts/ticks mode



SMPTE mode



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 variable according to the Frame Rate selected under menu: Settings|Time Code Format.

However, sample accuracy is often required when editing digital audio.

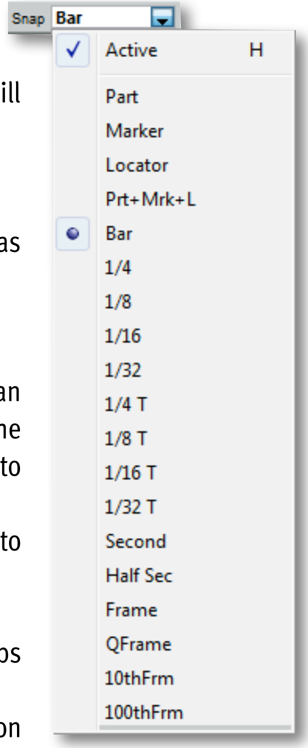
If **SMPTE time** is selected (menu: Settings|Time Axis), the mouse position readout employs a format that makes sample accurate editing with the mouse easy. It is necessary to use the highest zoom in level so that each pixel on the screen shows one single sample, and the Snap function needs to be inactive.

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

BACKGROUND INFO:
The actual number of samples within each frame depends on the Sample Rate and SMPTE Frame Rate selected. The table 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.

Number of samples per frame						
	24fps	**25fps**	**29.97fps**	**29.97df**	**30fps**	**30df**
22.050 Hz	918.75	882	735.74	735.74	735	735
32.000 Hz	1333.33	1280	1067.74	1067.74	1066.66	1066.66
44.056 Hz	1835.66	1762.24	1470	1470	1468.53	1468.53
44.100 Hz	1837.5	1764	1471.47	1471.47	1470	1470
47.952 Hz	1998	1918.08	1600	1600	1598.4	1598.4
48.000 Hz	2000	1920	1601.6	1601.6	1600	1600
64.000 Hz	2666.66	2560	2135.47	2135.47	2133.33	2133.33
88.112 Hz	3671.33	3524.48	2940	2940	2937.06	2937.06
88.200 Hz	3675	3528	2942.94	2942.94	2940	2940
95.904 Hz	3996	3836.16	3200	3200	3196.8	3196.8
96.000 Hz	4000	3840	3203.2	3203.2	3200	3200

Snap [H]



The Snap function defines a global time-grid for almost any editing and positioning function. With Snap active, any time position where Markers, Left and Right Locators and Current Locator will automatically "snap" to, will be the next point in the selected grid.
Also edits will be performed on the closest "snap point".

For instance, when Snap is set to Bar, a cut will be made only at a full bar, no matter if the mouse was exactly on a Bar boundary or slightly left or right.
Moving a part will always place its beginning exactly to a full Bar.

This allows fast and accurate editing without the need to zoom in. However, if you need to cut an audio Part between the third beat of the fourth bar and the second beat of the fifth bar and copy the resulting new Part to the third beat of the nineteenth bar, you need to set the snap value to 1/4 to make your cuts sample accurate, even if you position the Cut tool roughly.
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.

While working with picture, the Snap setting of one frame can be used to quickly place audio clips exactly as required.
The Snap function is also useful for positioning the Locators for punch in/out recording exactly on Beats.

Clicking on the **Snap** Drop Down Box opens the list of possible Snap Values.
The first item allows activation/deactivation of the snap function (or press the [H] key on the keyboard)
The currently selected snap value appears black in the Drop Down Box if the Snap function is active, or dimmed if it is inactive.

Also the mouse position readout is determined by the snap settings (please see above). Hence the displayed mouse position shows the time code of a potential edit.
Certain editing tools, such as the **Move Vertical** or **Copy to Locator** tools are not subject to the Snap function; they are not "snap-sensitive". Others (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.

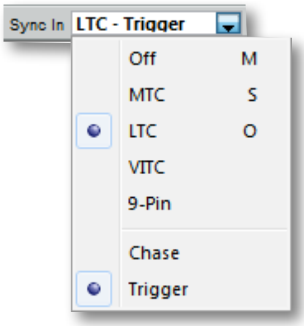
If the selected snap value is "Part", "Marker", "Locator", or "Prt+Mrk+L", then the beginning and end time positions of all Parts, any embedded snap points (as described in the "Snap Point Edit tool" of the "Editing Tools" section in this Chapter below), the Markers, the Left, Right, and Current Locators, or all of these together will respectively act as snap points whenever a snap-sensitive edit or operation is performed close enough to one of them. 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**, the end as well as the beginning of the edited Part will snap.
- Please also read about the **Snap Point Edit tool** in the section **Editing Tools** in this chapter.

Sync In [M], [S] or [O]

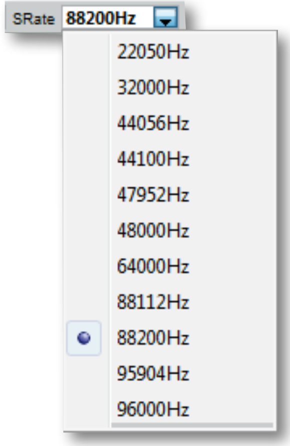


The Sync In box displays a list of "Synchronisation Slave" modes, also found in the menu: Settings|Sync In. Available options are: Off, MTC, LTC, VITC, and 9-Pin.

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 U1.
In addition the 9-Pin option is only available if "Controller", "Synchro" or "Layback" is selected in the menu: Settings| 9-Pin operation| Mode.
In Chase Slave or Trigger Slave modes, SSL Soundscape V6 will only respond to incoming Time Code if it is in Play mode or Record mode, i.e., if the Play or Record button has been clicked, unless the "Auto Play/Stop when slave syncing" option is enabled in the Sync Setup window (menu: Settings|Sync Setup).

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

Sample Rate

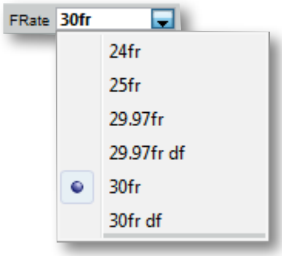


Clicking the SRate Drop Down box will display all available Sample Rates. Clicking on an entry will set Soundscape to operate at the corresponding Sample Rate. The Sample Rate setting can also be specified in the menu: Settings|Sample Rate.

When "Master Clock" is set to an external input in the Settings|Master Clock menu, the Sample Rate readout becomes a Display to show the Sample Rate being received. The Display will appear dimmed indicating that the Sample Rate selection currently 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.

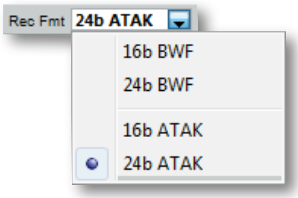
Frame Rate



Clicking the arrow to the right of the FRate box will open a list with available Frame Rates.

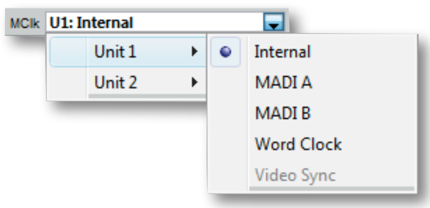
This parameter can also be set in the menu : Settings| Time Code Format.

Record Format



The Record Format box shows Soundscape's current recording file format and bit resolution. Clicking on the Rec Fmt Drop Down Box allows to selecting between 16 or 24 bit BWF broadcast wave file and 16 or 24 bit ATAK SSL Soundscape audio take format. This option is also available in the Settings menu. **Changing the Format will only affect newly recorded or processed audio Takes.**

Master Clock



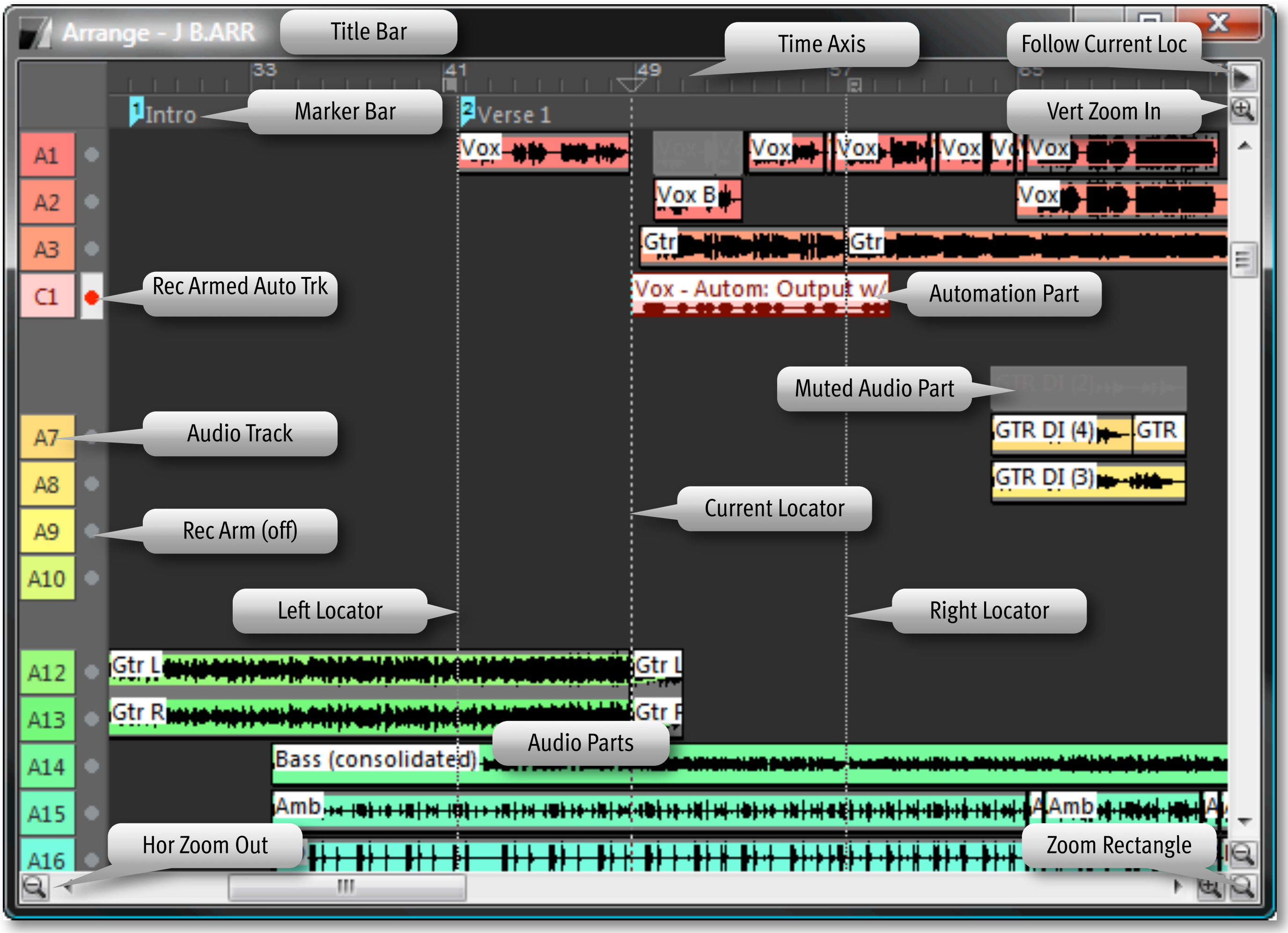
The Master Clock box shows the current master clock settings. The clocking options are shown for each unit present in the system. This option is also available in the Settings menu.

L<->R Readout

The L<->R readout shows the current time interval between the Left and Right Locators. The value is displayed in either bars/counts/ticks or SMPTE+sample extension according to the setting chosen for Time Axis in the Settings menu.

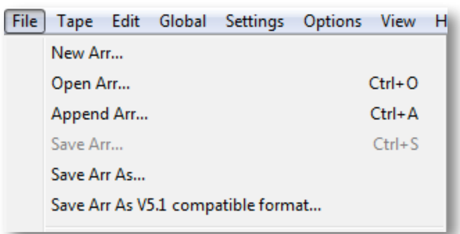
Arrange Window

The Arrange window is where most of the real work happens. It shows the layout of your Arrangement, with up to 256 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, that 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 a physical track (for audio Parts) or automation track (for automation Parts). Therefore muted audio Parts will not be heard and automation Parts will not control Mixer Elements when the Arrangement is played back.

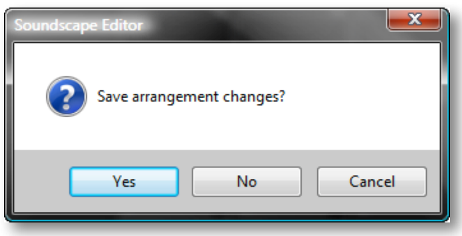


Create, Open and Save an Arrangement

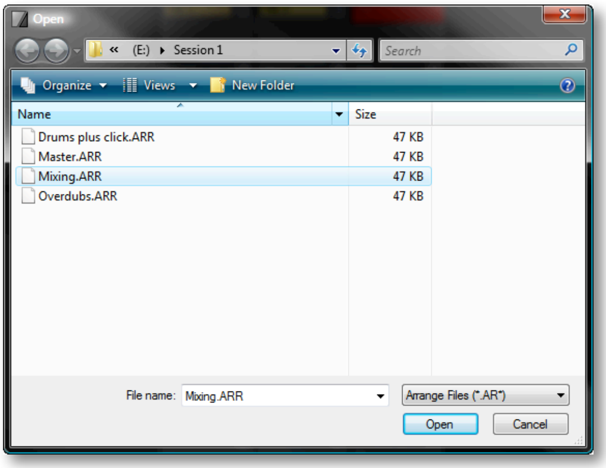
New Arr...
will create a new, empty Arrangement window.



If a currently opened Arrangement has been edited, you need to first commit to save or lose the changes or Cancel the operation completely and revert to the active arrangement.



Open Arr [Ctrl]+[O]
The Standard Windows File Browser appears.



If a currently opened Arrangement has been edited, you need to first commit to save or lose the changes or Cancel the operation completely to revert to the active arrangement. (see above)

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

Append Arr [Ctrl]+[A]
Append Arrange adds a saved .arr File to the currently opened Arrangement at the Current Locator position, so the Current Locator should be placed at the desired song position before using this option.

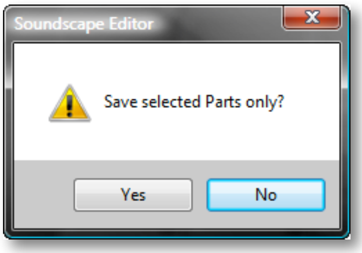
If Part overlaps are not allowed (menu: Settings|Preferences|Arrangement|Overlapping Parts (for new edits)) any Parts in the appended Arrangement which would "collide" or overlap with Parts in the current Arrangement, will not be loaded.

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

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 like 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 when saving the Arrangement.

Save Arr As
Saves the active Arrangement using a standard Windows Dialogue.

Saving selected Parts as an Arrangement
If any Part is selected in the Arrange Window when you click "Save Arr as" in the menu, a Dialogue Box will appear:



This function essentially allows you to only partially save an Arrangement and therefore create Multi Channel pieces, that can be used in other Arrangements using **Append Arr**.
Very useful for recurring Jingles, Openers or Sound FX.

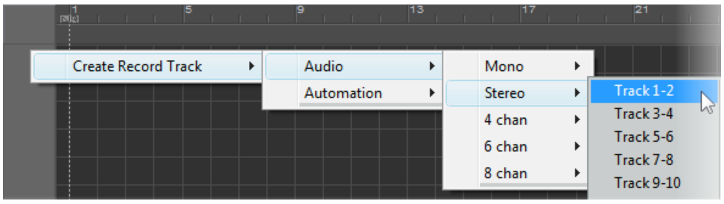
Click NO if you want to save the whole Arrangement.

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

Record Tracks

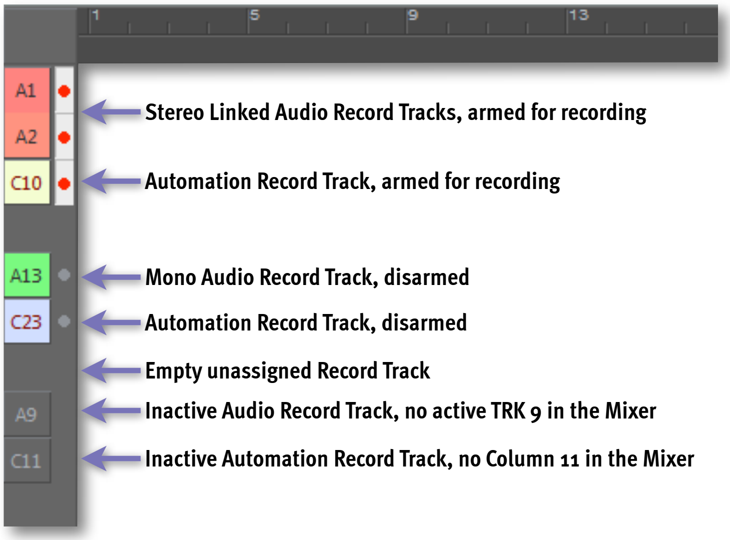
The Record Track Column allows you to determine the virtual track position, record loop stack size and track arming status for the physical track(s) to record to. Any record track inserted in the Record Track Column can also determine the default output track assignment for the corresponding virtual track when dragging Takes from File Manager 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 with submenus allowing you to 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 single mono tracks, stereo pairs or multichannel track groups. For automation record tracks, please note that you select the Column Slot Number inside the Mixer Window you want to automate.

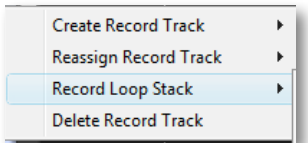
There is also no fixed relation between Track I/O Assignments and the Mixer Column Number in Soundscape V6, hence Track 1 Audio (A1) could be connected to Mixer Column 10, making it necessary to create automation for this Audio Track on Automation Track C10.
Baffled? Confused?
The good news is: You can assign Audio Track 1 to virtual Track 1 and Mixer Column 1...if you like to keep it simple;-)



Once created, a record track appears as a colour-coded slot in the Record Track Column, with a round “track arming button”.

The rectangle will be dimmed and the track arming button will be absent for an audio record track, if there is no corresponding track insert in the Mixer, and also for an automation record track if there is no corresponding mixer column.

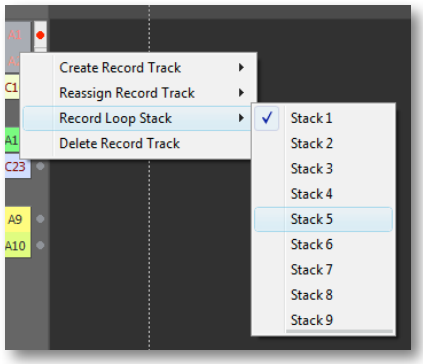
NOTE: A mono Audio Record Track cannot be used and will remain dimmed if it is connected to a Stereo Track Insert in the Mixer.
A stereo Audio Record Track however will be enable both Mono Track Inserts in the Mixer.



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), but you can also reassign an existing record track (i.e., change its Track number), create a record loop stack (for loop recording) or delete an existing record track.

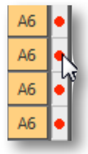
If you want to replace an automation track with an audio record track simply select **Create Record Track->Audio->Trk...** in the menu. If you select Reassign Record Track, you can change a track’s number but not its type.

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 existing record track or record loop stack can be moved to a new position in the Record Track Column by clicking and dragging it with the left mouse button. Several record tracks can also be selected at the same time by holding down the [Ctrl] key and clicking them one by one as required, or by holding down the [Shift] key and clicking any two record tracks to select them along with all the other record tracks between them. Multiple selected Record Tracks can be dragged as a group to new virtual track positions, or deleted.

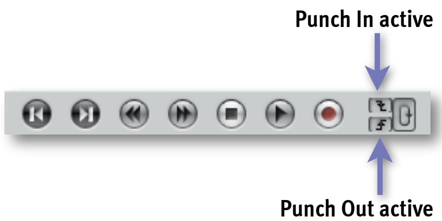
If you drop a record track or record loop stack on an existing record track, this existing record track disappears.
The Undo function can be used to reverse these changes.
A record loop stack can be armed for recording by pressing any single one of the track arming buttons within that stack.



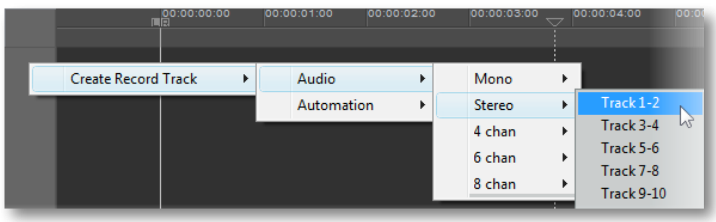
Recording Audio

1. Soundscape V6 opens **Soundscape def.mix** by default. If your Mix Window is empty you need to open a Mixer for recording (menu 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) by clicking either or both buttons in the Tape Transport Bar.
If the Auto Punch In function is inactive, the Current Locator determines the Position where recording is starting (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.

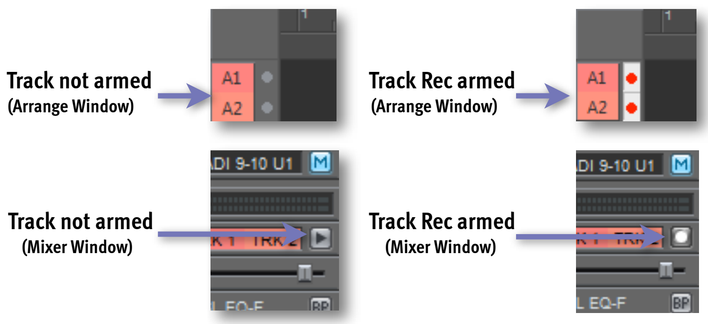


3. Right-click on a free slot in the Record Track Column. A menu will appear that lets you select mono or stereo and which tracks to record onto. Release the mouse button on the number of the track(s) you wish to record onto. If you are using **Soundscape def.mix**, select stereo and choose "Track 1-2":



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 are colour-coded the same way 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 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.

- Pressing the [*] key arms all the available audio record tracks
- Pressing a numerical key followed by the [*] key only arms the corresponding audio record track e.g. pressing [5][*] arms track 5.
- Pressing two numerical keys separated by the [-] key and followed by the [*] key arms all audio record tracks from the first to the second number, e.g. pressing [3][–][8][*] arms all existing audio record tracks from track 3 to track 8.
- Pressing the [/] key disarms all previously armed audio record tracks.
- Pressing a numerical key followed by the [/] key only disarms the corresponding record track, e.g. pressing [5][/] disarms track 5.
- Pressing two numerical keys separated by the [–] key and followed by the [/] key disarms all audio record tracks from the first to the second number. For example, pressing [3][–][8][/] disarms all existing audio record tracks from 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.

Independent 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 is generally not crucially important.

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

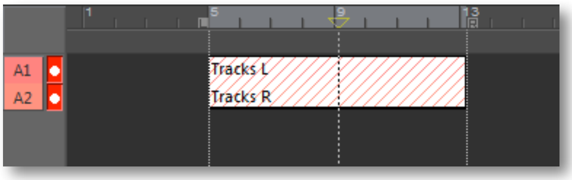
WARNING: Be very careful not to cause the input to clip while recording, as with all digital systems, this can result in very high distortion levels, especially for full-scale Input longer than half a dozen of samples. Digital Systems do not have a concept of "Head-Room", it is therefore a good idea 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 is checked in the Record Setup window, (menu: Settings|Record Setup), 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, using the key commands described above). 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 menu: Settings|Preferences|Arrangement is set to "Not allowed", existing Parts underneath the new recorded Part (i.e., on the same virtual track) will be cut and deleted. If it is set to "Allowed", any new recorded Parts are stacked on top of existing ones.
The default Recording time for manual recordings can be changed in **Record Setup** (menu: Settings) and using the Record Setup window. By default this will be set to 0 minutes, allowing unlimited recordings, until you press Stop.

7 If Auto Punch In is active, SSL 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 under menu: Settings|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" chapter).

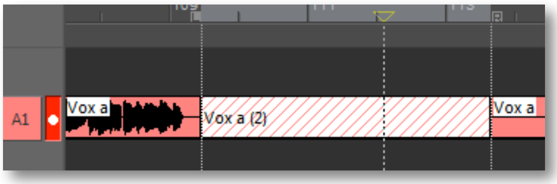
NOTE: If the recording is interrupted (i.e. if there is a power failure or PC crash), the recorded Takes will be saved and only missing the last couple of seconds before the PC stopped working.

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 (i.e. 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 is automatically 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).

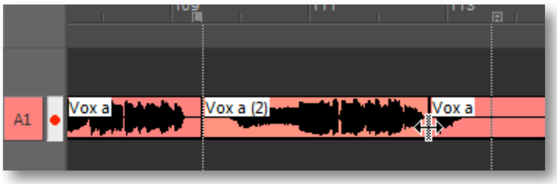
The pre-existing audio is not deleted from the Disk when you record in Auto Punch In/Out mode, so even if the Auto Punch In and Punch Out points (Locator positions) were not defined accurately, the edit point can easily be adjusted later (provided that there is enough audio material recorded).

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 finetune the I/O points to make the edit between original and overdub seamless.

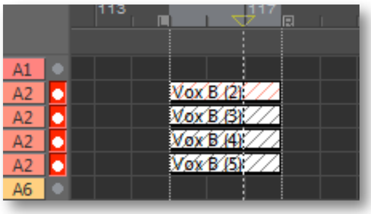


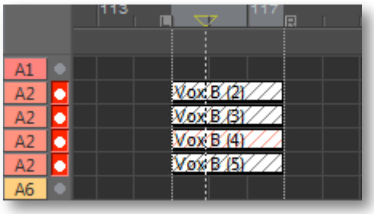
Loop Recording and Loop Stacks

“Loop mode” is made active by pressing the Loop button in the Tape Transport Bar:

If recording is started with the Loop mode and Auto Punch In/Out activated, Soundscape drops into recording when the Current Locator reaches the Left Locator.
When the Current Locator reaches the Right Locator recording starts again at the Left Locator (unless “Auto stop” is set to “at end of recording loop stack” in the menu: Settings|Record Setup).
If the Stop button on the Tape Transport (or the [Down Arrow] key) is pressed close before the Right Locator, the last recorded Take is preserved. Otherwise it is overwritten during the next pass.
If recording is started with the Loop mode made 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.



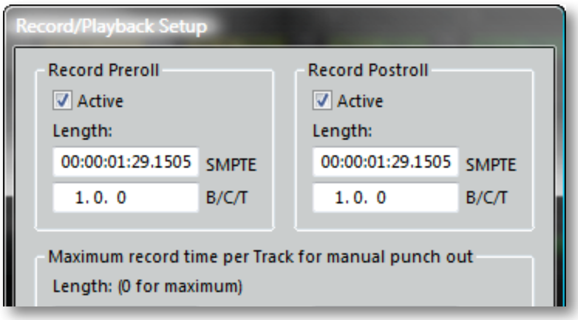


NOTES:

- An existing record loop stack in the Record Track Column will only be taken into account if the Loop button is pressed.
- If SSL 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 and the sequencer should 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 before the Left Locator, according to the specified preroll length and only 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, according to the specified postroll length, before either:

- stopping (if “Auto stop” is set “At end of recording” and unless Loop mode is active).
- returning to the preroll start 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”).

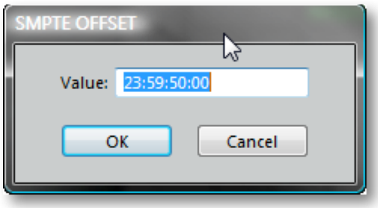
NOTE: The preroll setting only has an effect if Auto Punch In is active, and the post-roll setting only has an effect if Auto Punch Out is active.

Recording while Slaving to external Timecode

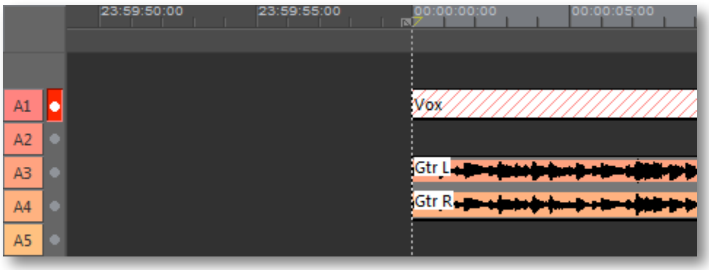
Currently external Time Code synchronisation is only available when a Soundscape 32 unit is present in the system.

In the “Synchronisation Slave” modes (i.e., when Sync In is set to MTC, LTC, VITC or 9-Pin, as shown in the Status Bar or under menu: Settings|Sync In), 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. This takes a variable amount of time, depending on the number of tracks to be played back, 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 menu: Settings|Sync Setup. The default is 2500ms, but any value can be set between 100ms and 4000ms. Since hard disk performance is always improving, extremely low settings are generally sufficient nowadays, especially for simple material.
However, when recording with SSL 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 (e.g., 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 SSL 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 a case, open the SMPTE Offset window by clicking “SMPTE Offset“ under the Tape menu or the value in the SMPTE Offset readout, enter a value (e.g., 23:59:50:00 for 10 seconds of empty time before 00:00:00:00 is reached in the Arrangement), and click **OK.**



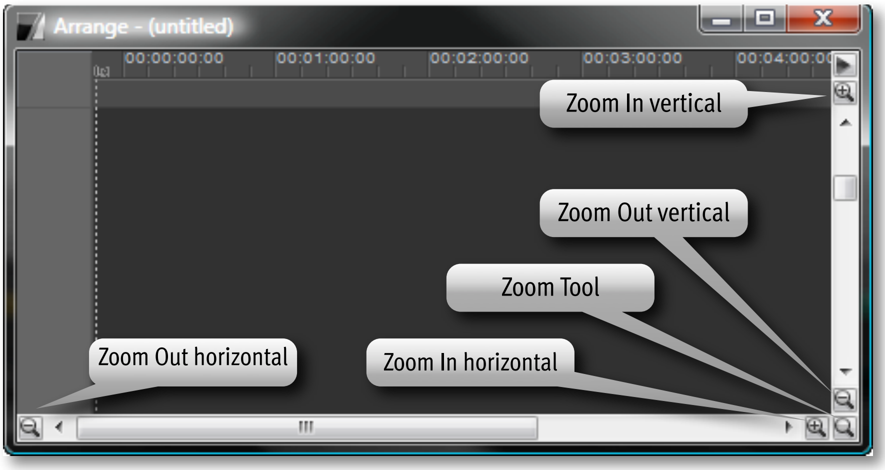
Place a Locator at 00:00:00:00, or at any other position in the Arrangement where the first Part should start, select all the Parts in the Arrangement (by clicking “Select all Parts” under the Global menu or pressing [Shift]+[A] - this way you can be sure that all existing Parts are actually selected, including any Parts that are not currently visible), and use the “Move To Locator” tool (as described in the “Editing Tools” chapter) 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 this (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, and recording will 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...).

Zoom

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 V6 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 inside 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.

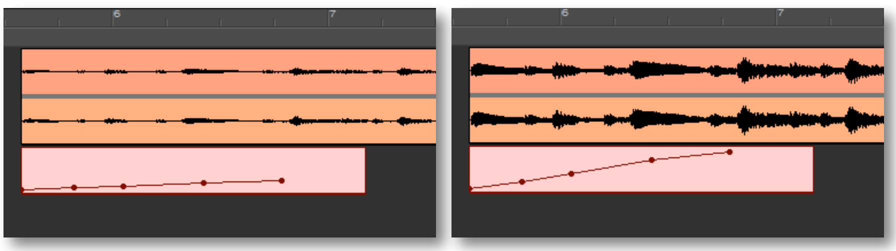
Horizontal Zoom

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 (as the zoom level increases it can be necessary to readjust the Current Locator position).
It is often preferable to use the Zoom tool when accuracy is required.
If the Current Locator is not visible when zooming in horizontally, the visible centre of the Arrangement stays fixed while zooming in.

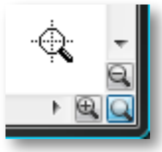
Vertical Zoom

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).

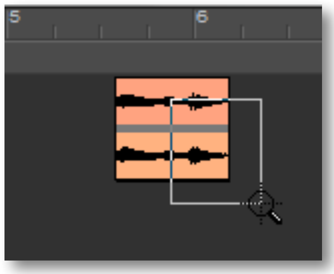
Right-clicking the Vertical Zoom In/Out buttons zooms only the Waveform or Automation Curves inside the Parts and allows magnifying the vertical amplitude of the audio waveforms and automation curves. This does not alter the audio or automation data, but allows to see low level waveforms and automation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



Zoom Tool [Z]

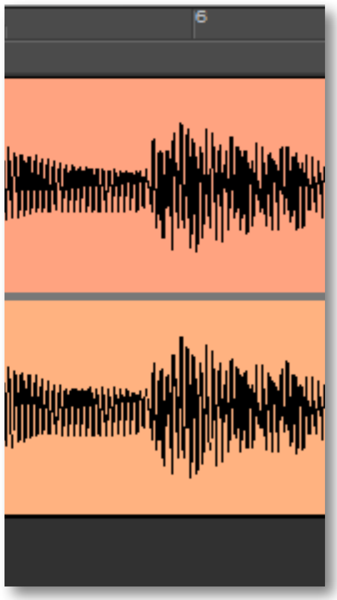


Clicking once on the Zoom Tool (or pressing the [Z] key) will start the Zoom Tool and turn the mouse pointer into a magnifying glass when hovering over the Arrange Window.



Now 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 reverts to it's previous shape and function, the Zoom Tool is a temporary 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: it is possible to draw a selection box that extends beyond the current view of the Arrangement. The Arrange window will be scrolled automatically in the required direction when necessary, therefore 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 (2x [Z] & [Shift]+2x [Z])
By double clicking the Zoom Tool (or hitting 2x [Z]) 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 "Macro and Micro" Views, or between Project Overview and Track Detail View.
Soundscape V6 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.

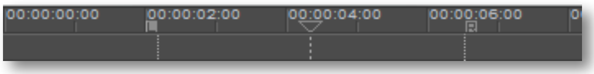
Scrolling the Arrange window
The Arrange window can be scrolled using the Scroll buttons in the usual Windows way.
It can also be scrolled using the following key commands:

- 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 menu: Settings|Time Axis.

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 each single tick.

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.

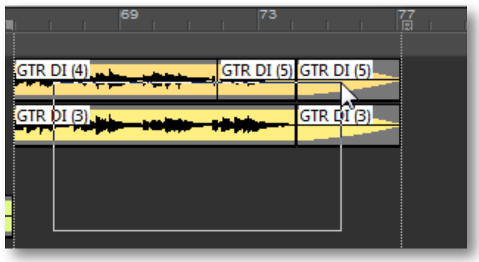
NOTE: The snap value can be set to quarter, 10th, or 100th frame, and at maximum zoom level with the Time Axis set to SMPTE time, the mouse position readout shows single samples for ultimate precision while editing.

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 the 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 (e.g., copying, cutting etc.) does not affect the corresponding Takes on the Disk.

Altering the Audio Data by processing a Part (e.g., normalising, pitch shifting, or reversing it) will create 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 (e.g., 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 that 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 the [Shift] or [Ctrl] key 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.

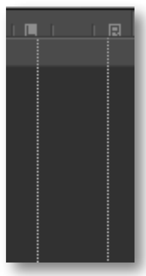
A Part can be individually selected or deselected by clicking on it while holding down the [Ctrl] key on the computer keyboard. This can be useful for example to deselect a Part in the middle of group of selected Parts, or to build a group one Part at a time.

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 the menu: 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 affect the operation of certain editing tools (e.g., Mixdown tool, Crop tool...), they are used by some of the Global functions (as described in the **Global Menu** Section of the Menu Reference chapter), they are used as snap points for certain snap settings, they can be used for automation snapshot recording, etc.

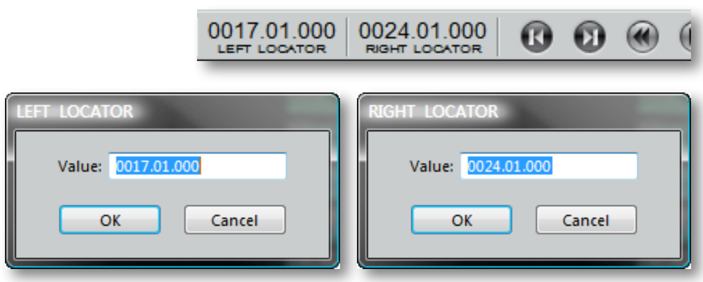
The time interval between the Locators is displayed in the L<->R Display inside the Status bar.

Setting the Left and Right Locators

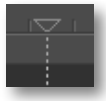
To position the Left or Right Locator, click at the required position in the Time Axis with the corresponding mouse button. The Locator will be set to the mouse position, or to the closest snap point if the Snap function is active. A Locator can also be moved along the Time Axis by clicking and dragging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

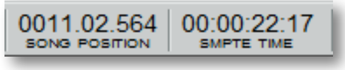
Alternatively, you can click the Left or Right Locator Display in the Tape Transport Bar to alter the time position in a dialog box. Depending on the Time Axis Setting SMPTE Time or Bars/Beats/Ticks are diplayed.
You can type [Space] instead of [.] or [:] and leading zeros are not required. The time positions entered in the dialog boxes for the Left and Right Locators will appear in the Displays as 00017.01.000 and 0024.01.000 .



Current Locator



The Current Locator shows the current play back or Song Position in 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.

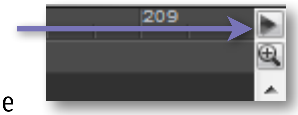
The Current Locator can be positioned in several ways:

- 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. 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 or altering the value in the dialog 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 menu: Settings|Preferences|Arrangement.

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).

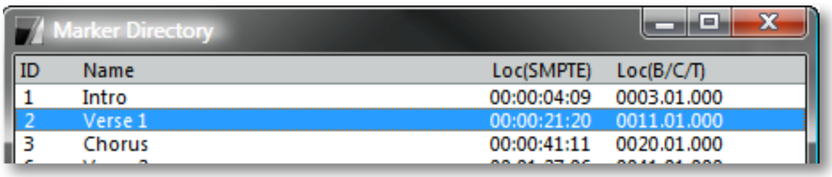


Disabling Follow Current Locator is especially useful while focussing on a particular edit point and working at a high zoom level.

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 [W]

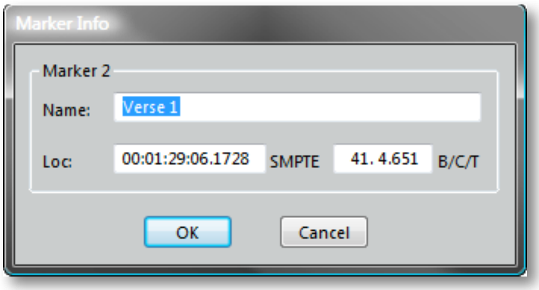


The Marker Directory can be shown or hidden by ticking or unticking “Marker Directory” under the View menu or by pressing the [W] key. This window shows a text listing of Markers, with their ID number, name and time position in both SMPTE and bars/counts/ticks.

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, but you can type any ID number from 1 to 999 before pressing [Insert], so that the ID of subsequently inserted Markers will be incremented from that number.
This makes it easy to identify different sections of an Arrangement with the Marker ID numbers (e.g., 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, and the Marker Directory remains "on top" when you insert Markers. So you could 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. 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 use the mouse to move a Marker to a new position by clicking with the left mouse button on the Marker and dragging it. Active Snap settings will determine available Marker positions when dragging with the mouse.

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 using the [Shift] and [Ctrl] keys. Holding the [Shift] key allows selection of a range of markers by clicking the 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 SSL Soundscape V6 Window contains the above mentioned Displays for Locator/Song Position, the "Tape" Transport Section and buttons for Auto Punch In/Out and Loop Playback/Record and also Displays for musical Time SIgnature (SIGN), Tempo and SMPTE Timecode Offset.

Tape

To Beginning [Home]
Current Locator jumps to the beginning of the first Part(s) in the Arrangement.

To End [End]
Current Locator jumps to the end of the last Part(s) in the Arrangement.

To Beginning and To End while holding [Shift] or [Ctrl] keys
By additionally holding down the **[Shift] key** 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 operation will interrupt playback or recording.
It is particularly useful when working with video and Snap set to Frames.

By additionally holding down the **[Ctrl] key** the Current Locator will move to the nearest previous or next Part beginning or Part end. Using this function will interrupt playback or recording.

Rewind and Fast Forward [Arrow Left] and [Arrow Right]
Rewinding or fast forwarding is active as long as the buttons are pressed.
If REW/FFW is pressed during Play mode, Playback will resume as soon as the button is released.

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

Technically there is no need to rewind or fast forward in a hard disk based digital recording system, yet these functions resemble familiar workflows and are actually useful, for example while in Play mode, to skip a short section of the Arrangement with a quick mouse click.

Stop [Arrow Down]
If Soundscape is in Play or Record mode, clicking the Stop button 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 [Arrow Up]
Clicking the Play button will initiate playback from the current song position.
Clicking the Play button after typing a Marker number on the numerical keypad initiates playback from the corresponding Marker location.

Record [NUM +]

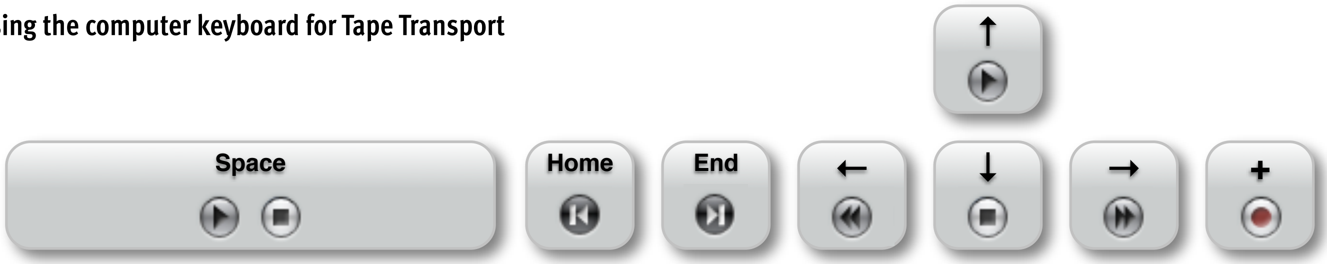
Clicking the Record button initiates recording from the current song position (if Auto Punch In and Pre-Rolls are not active).

Recording can be started while Soundscape is stopped (in this 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 (and playback). 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.

Stop

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

REW/FF

The Left 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 (menu: Settings|Preferences|General), pressing [Left Arrow] will rewind in 1x playback speed, providing a “Reverse Play” mode. Repeated key presses will increase REW speed incrementally , 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

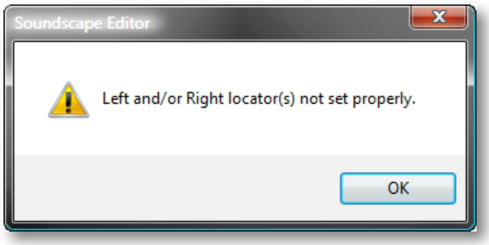
Clicking these buttons or typing the corresponding key commands toggles the Auto Punch In and Out modes and the Loop mode on or off.

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 following warning will appear.
Click **OK** and change the Locator positions.
If Loop mode is made active in Play mode, upon reaching the Right Locator the Current Locator 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.

Time Signature and Tempo readouts
The Time Signature and Tempo readouts are located at the right hand side of the Tape Transport Buttons.

Clicking either of them opens a dialog box in which the Time Signature or Tempo can be entered.

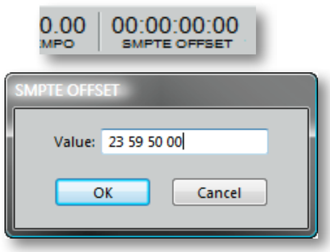
When entering the Time Signature, the numerator can be set to any value from 1 to 15 and the denominator can be 4, 8 or 16. If you try to enter a value outside these ranges the closest available value will be used instead. Leading zeros are not required and you can type [Space] instead of [/].

The Tempo can be set to any value between 30 and 240 beats per minute with an accuracy of two decimal digits (e.g. 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 V6 is especially important while editing with musical Snap values or working in Sync with a MIDI Sequencer. Changing Tempo or Time Signature, however, 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 dialog 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 the Time Code from an external device.

For instance, 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.

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 (e.g., 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 [Ctrl]+[1]

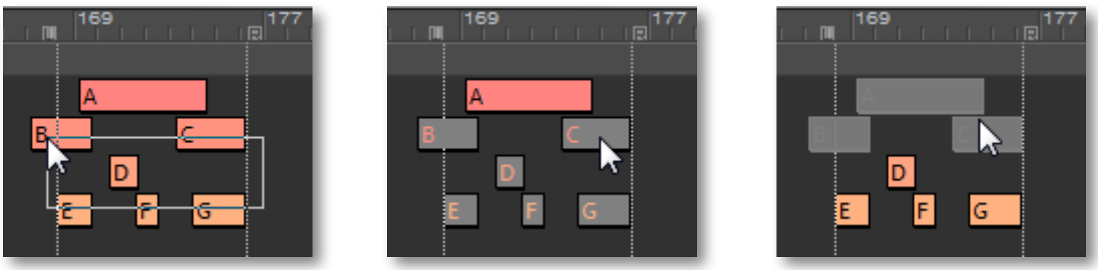
This tool works on audio and automation Parts.

The Track Assign tool allows you to select which physical audio or automation track and which Part (or group of selected Parts) are assigned to each other.

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 menu: Settings|Preferences|Arrangement:

- 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 (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 remains 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 screenshot). 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 **Chapter 5 › Settings Menu** for complementary information.

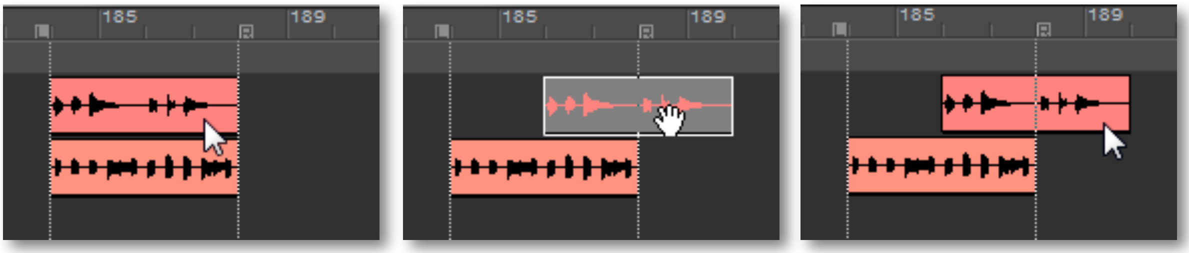
Move tool [Ctrl]+[M]

This tool works on audio and automation Parts.

The Move tool allows any Part (or group of selected Parts) to be moved to any position in an Arrangement.

To move a single Part

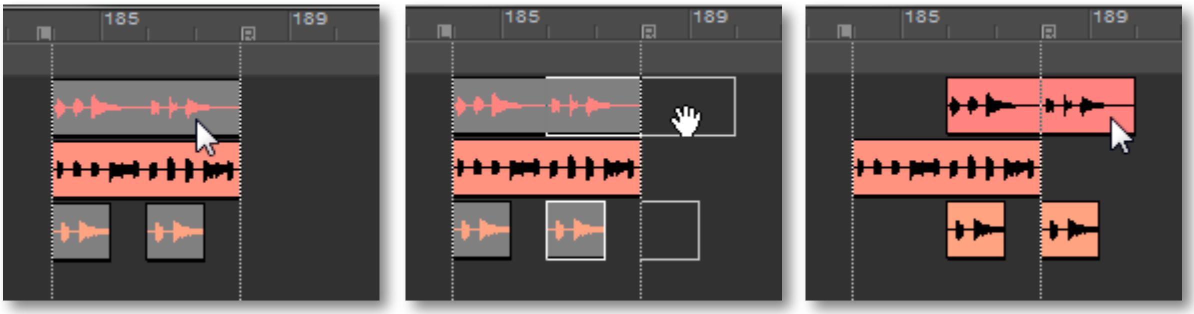
Once you have assigned the tool to either mouse button, click and hold any Part and drag it to the desired position. If the target position is outside the currently visible section of the Arrange window, hold the dragged Part on any Arrange window border and the Arrangement will auto-scroll in the corresponding direction. Release the mouse button to drop the Part at the pointer position. If the snap setting 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.

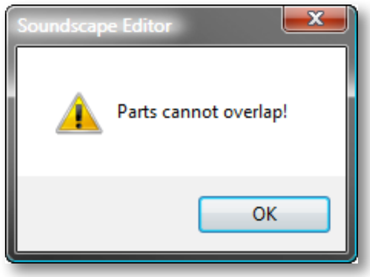
To move a group of selected Parts

Click and hold any Part within a group of selected Parts to drag the whole group to a new location. If the "snap" setting is active the Part you initially clicked on will jump to the nearest snap point when the mouse button is released, but the overall layout of the group relative to that Part will remain unchanged with respect to timing.



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 menu: Settings|Preferences|Arrangement. This applies whether the Part is moved individually or within a group.

If the moved Part or group is overlapping an existing Part when you release the mouse button, several things could 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 V6 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 [Ctrl]+[C]

This tool works on audio and automation Parts.

The Copy tool 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.

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

Move & Copy Vertical tools [Ctrl]+[Shift]+[M] (or [C])

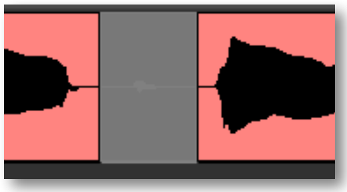
These tools works on audio and automation Parts.

They only allow edits in the vertical direction (i.e., from one virtual track to another). This ensures that no change in timing occurs.

Cut tool [Ctrl]+[X]

This tool works on audio and automation Parts.

The Cut tool 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 the snap setting 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 thereafter 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 the snap setting 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 shown on the right):

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** [Ctrl]+[Alt]+[M] (or [C], or [X])

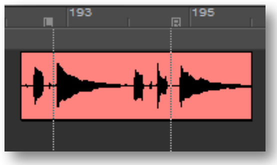
These tools work on audio and automation Parts.

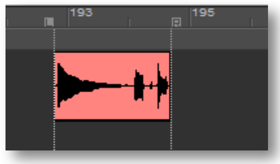
The Move to, Copy to, and Cut at nearest Locator tools allow Locator based editing. The Current Locator, Left Locator, or Right Locator, (whichever the mouse pointer is closest to), determine the position of a move, copy, or cut operation. With the Move tool or Copy tool, the beginning or end of the 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 “on the fly” (while playing), or by entering the Locator Position as a value in a Tape Transport Bar readout.

 **Crop tool** [Ctrl]+[3]

This tool works on audio and automation Parts.

The Crop tool 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.





 **Solo tool** [Ctrl]+[H]

This tool only works on audio Parts.

The Solo tool can be used to play the Parts (or stereo/multi-linked Parts). To solo an active Part (not muted) click on it or in front of it and hold the mouse button down. The Part will be played from the mouse position and 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 or 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 menu: 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.

Solo Part Begin/End tool [Ctrl]+[Shift]+[H]

This tool only works on audio Parts.

The Solo Part Begin/End tool 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/is placed within a Part, you can solo up to a Locator or away from a Locator, or from one Locator to another. 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.

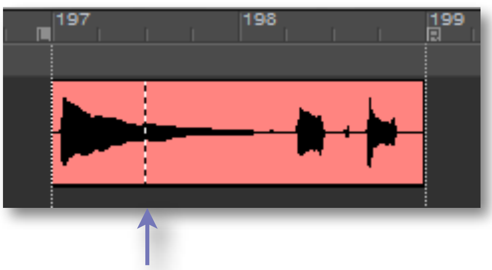
The tool also takes embedded snap points into account, as described in the **Snap Point Edit tool** section below.

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

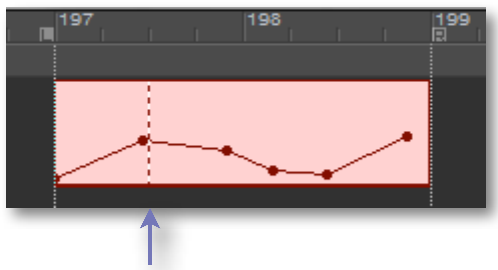
Snap Point Edit tool [Ctrl]+[Shift]+[D]

This tool works on audio and automation Parts.

Clicking an audio or automation Part with The Snap Point Edit tool will create an "embedded" snap point within that Part, at the time position where the mouse pointer is located, or at the nearest snap point if the Snap function is active. The embedded snap point appears as a dotted line across the Part:



Audio Part Snap Point



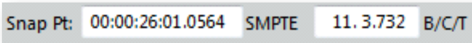
Automation Part Snap Point

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 the Snap function 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 an important point in an audio Take or an important event in an automation curve, which needs to be positioned precisely, e.g., on a particular frame when working with video, or on a particular beat when editing music.

- If the Snap function 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 (instead of 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 the Snap function 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 that it starts playback from, or like a Part End or locator that it stops playback 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.



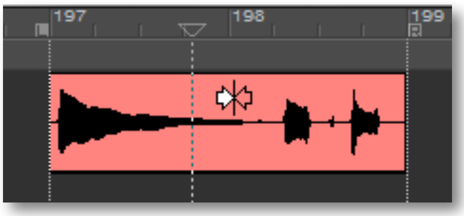
When editing the value using the computer keyboard, leading zeros are not required and you can type [Space] instead of [:] or [.].

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 [Ctrl]+[B] & [Ctrl]+[Shift]+[B]

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. This is similar to moving the tape across the tape heads by rocking the reels of a tape machine, to find a precise edit point.
With the Solo Scrub tool selected, pointing at an audio Part or at a pair of stereo/multi-channel linked Parts, clicking and holding the mouse button down 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 center line. Moving the tool away from the Current Locator in either direction will trigger playback in that direction.
The selection made under menu: Settings|Preferences|Arrangement|Scrub tools behaviour 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 is again 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 (menu: Settings|Preferences|Arrangement).
The Multitrack Scrub tool works in the same way as the Solo Scrub tool, except that it always 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.

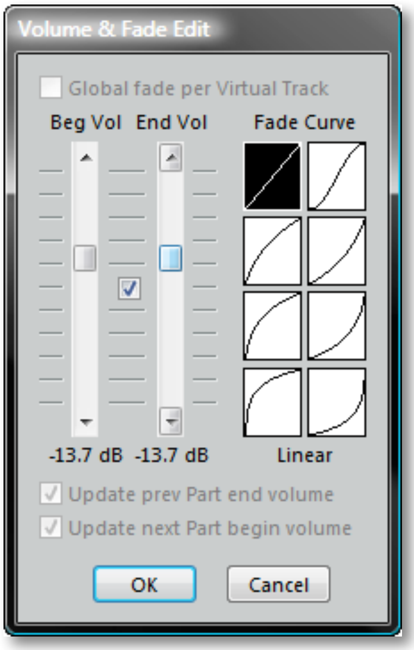
Part Volume and Fade tools

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. Part volume is indicated on-screen by the vertical amplitude of the coloured section of a Part, if the **Show Part Volumes** option is enabled (menu: Settings|Preferences|Arrangement). However, due to screen resolutions, volumes are only displayed from the third vertical zoom level on.

Volume & Fade Edit tool [Ctrl]+[V]

Select the tool and click on an audio Part. The Volume & Fade Edit window will appear:



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, but as the faders follow an audio taper law for dB versus on-screen movement, the distance between the faders on-screen will not be constant as the faders are moved.
To edit the beginning volume or end volume individually, make sure that the check box 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 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 respectively 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 that it reduces volume by a fixed amount per second. This means that after half of the fade time, the level is half of the original (-6dB). After 3/4 of the fade time, the level is 1/4 of the original (-12dB), etc. This means that the rest of the fade (to -96dB) takes place in the last 1/4 of the time. 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** [Ctrl]+[E] & [Ctrl]+[F]

The Fade In and Fade Out tools change the beginning volume to -96dB or the end volume to -96dB respectively without affecting the other volume setting. The curve setting is left unchanged.

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. i.e. you will lose any volume contours that may have been previously defined!

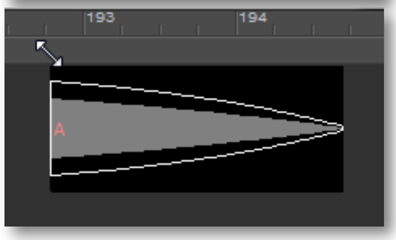
 **Volume Trim tool** [Ctrl]+[Shift]+[V]

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 (i.e., a volume curve defined using the Volume Trim tool will have the shape selected in the Volume & Fade Edit window).

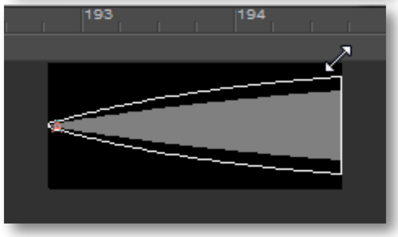
Basic operations

Click and drag vertically with the mouse in the centre of a Part to change both its beginning and end volumes, so that the slope is unchanged:

Click and drag on the left side of a Part to change only its beginning volume:



Click and drag on the right side of a Part to change only its 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. If you click and drag in the centre of the Part, as 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 the operation of the Volume and Fade tools 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, starting at the beginning volume level and finishing at the end volume level, which approximates to the chosen curve (shown in black), will be made across all selected Parts on each virtual track. Each individual Part will then have this curve selected, so for a closer overall approximation to the curve, select all the Parts again, use the Volume & Fade Edit tool, turn off the global check box and select 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 change the Link Faders 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, then use the Volume & Fade Edit tool. Make sure that the global check box is unchecked and that the volume 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.

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. This could be because the beginning volume of the first Part is at minimum (e.g., a fade in) and the end volume of the last Part is also at minimum (e.g., a fade out). Then both faders would be at -96dB and so the level of the track could not be reduced by say 6dB. The easy way to accomplish this change is to select all required Parts except the fade in/out ones 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, as then 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.

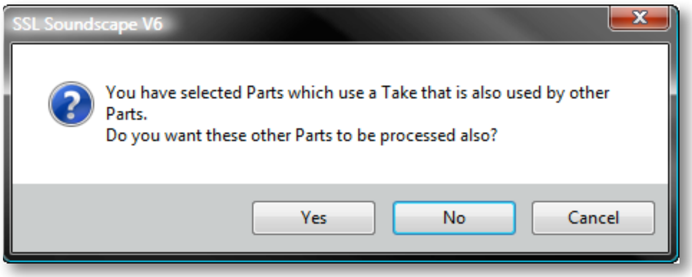
However, unless the edited Part has the linear curve shape selected first, the approximation to the overall shape required may not be as regular as expected, because the overall result will be, by definition, a succession of non-linear curves.
To change the curve setting of each Part to linear, first 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 [Ctrl]+[N]

This tool only works on audio Parts.

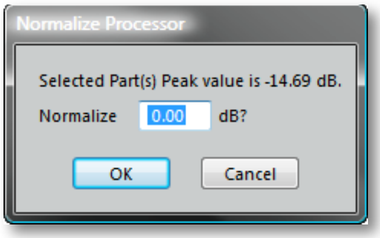
Normalising can be used to adjust the overall level of a Part or the level of a whole Take so that its peak matches a value of your choice. This process is used typically to boost a signal which was not recorded at a sufficient level. A new Take is generated to disk and renamed automatically (e.g., #1 or #2 is appended to the original Take's name).

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



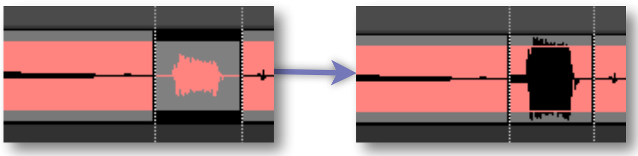
If you have selected any Part which is not the entire length of the Take on the Disk, or which references a Take also referenced by other Parts which you did not select, the following message will be displayed:

Click "No" to generate (a) new normalised Take(s) just for the Part(s) selected, i.e., 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.



If you wish to continue with the normalise process, enter the value of your choice for the new peak level and click "Ok", otherwise click "Cancel" and nothing will have been altered. (Please remember to include a minus sign when entering the desired 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.



NOTE: 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" and 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 and 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 [Ctrl]+[Shift]+[T]

This tool only works on one mono Part or stereo/multichannel linked group at a time.

The Time Stretch/Compress Process tool changes 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 [Ctrl]+[Shift]+[P]

This tool only works on one mono Part or stereo/multichannel linked group at a time.

The Pitch Shift Process tool changes 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 [Ctrl]+[Shift]+[S]

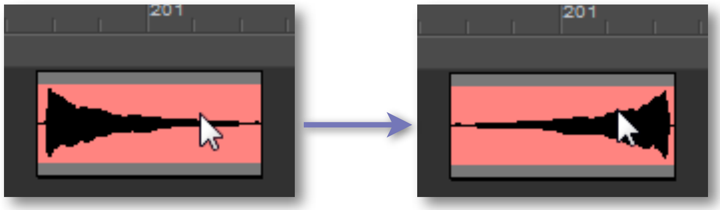
This tool only works on one mono Part or stereo/multichannel linked group at a time.

The Sample Rate Convert Process tool resamples 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 [Ctrl]+[Shift]+[R]

This tool only works on one mono Part or stereo/multichannel linked group at a time.

The Reverse tool generates a new "reversed" version of the Part (or stereo/multichannel linked group) that has been selected, along with the corresponding Take. It is similar to the process of playing 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 [Ctrl]+[Shift]+[N]

This tool only works on audio Parts.

The DC Removal tool processes 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 [Ctrl]+[U]

This tool works on audio and automation Parts.

The Mute tool allows any Part or group of selected Parts to be muted or de-muted by selecting the tool and clicking on that Part (or one Part in that group). Muted Parts retain the track assign data and de-muting, using the same tool, restores the memorised track assignments. 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 [Ctrl]+[G]

This tool works on audio and automation Parts.

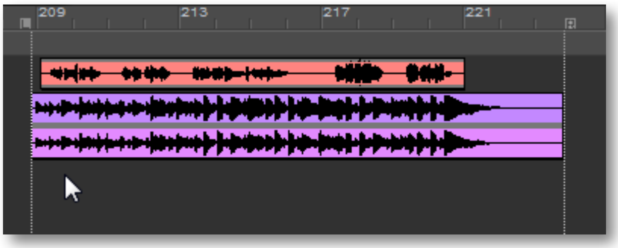
The Glue tool allows you to remove cuts that have been made to a Part in the Arrange window. To glue a Part, assign the tool to the left or right mouse button, point at the Part section which is immediately before the cut, and click. The cut will be removed and the glued Part will take on the parameter settings (i.e., 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 (i.e., continuous audio).

NOTE: If you want to create one Part from several Parts in order to use just one set of parameters for all the Parts, but the Parts use different Takes, then use the Consolidate tool to create a single new Take.

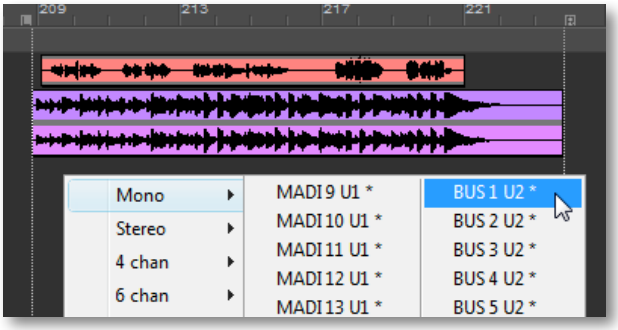
Mixdown tool [Ctrl]+[Shift]+[Y]

This tool creates audio Parts.

The Mixdown tool allows all the active tracks to be digitally mixed together to 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 and fade settings, 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 (i.e., 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).



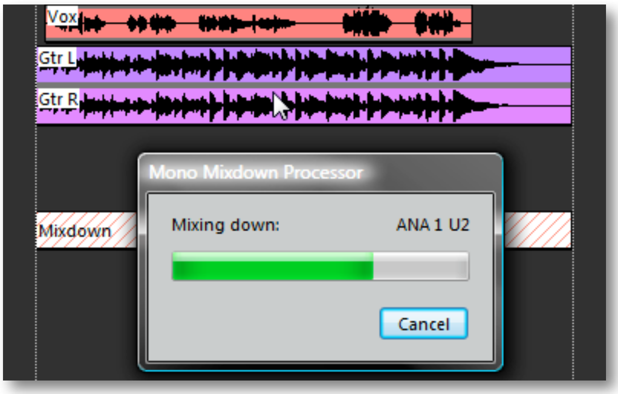
To Create a Mixdown, first position the Left and Right Locators around the region you wish to mix. The length of the new Take is determined by the Left and Right Locators, so place them accurately to avoid using more Disk space than is actually required. Once you are happy with the volume, pan, fade 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 mono, stereo or multichannel mixdown, and finally which output or bus in the Mixer shall be recorded from. The output selection is critical as all audio is routed through the Mixer, so only tracks routed to the selected outputs will be mixed down.

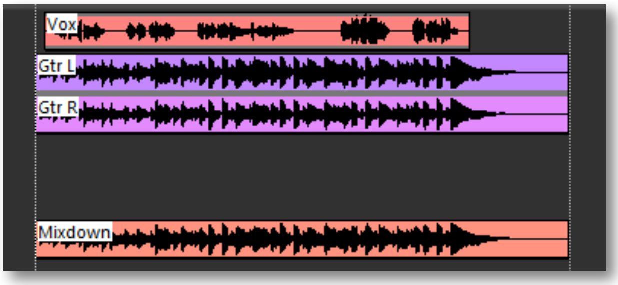
For example, you may have tracks 1 to 4 routed to outputs Bus 1-2 and tracks 5 to 8 routed to outputs Bus 3-4 in your Mixer. If so, selecting Bus 1-2 will only mix down tracks 1 to 4. It is a good idea to bus all of your tracks to a Master mixer column so they are all available from one output selection. This would allow you to control which tracks will be mixed by selecting or muting them in the Arrange window. See the Mixer chapter 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 following dialog box appears to show the progress and you will also hear the audio that has been selected between the Locators.

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



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

The Mixdown creates new audio data and saves it on the 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 the SSL Soundscape's Mixer. These signals could be inserted 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** [Ctrl]+[W]

This tool works only on audio Parts.

The Waveform tool is used to turn on and off the waveform display for selected Part(s) 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 the 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 for Parts that have a low volume setting, or at the end of a fade out.

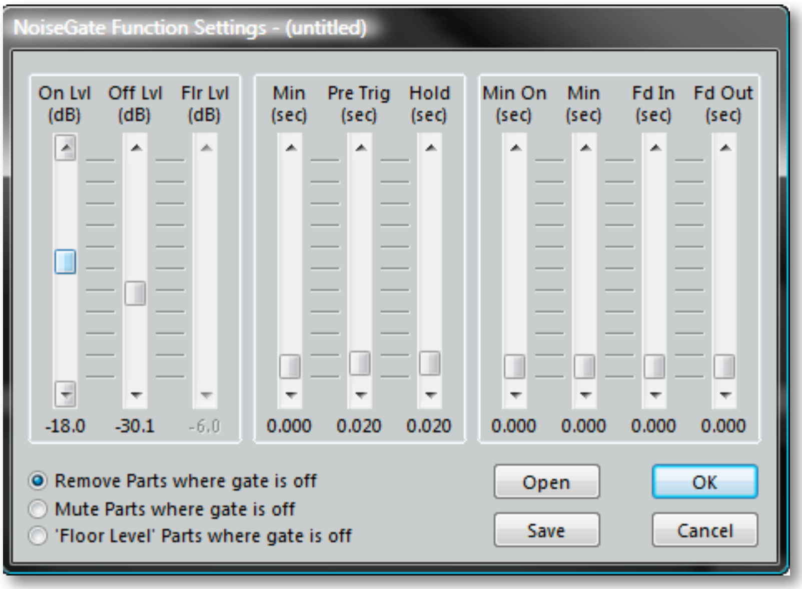
You can select whether waveform files will be saved or discarded upon exiting the program with the option **Keep all generated waveform files** (menu: Settings|Preferences|General).

Noise Gate tool [Ctrl]+[K]

This tool works only on audio Parts.

The post-processing Noise Gate function 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] can 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 10 noise gate parameters. This generates multiple smaller Parts, but leaves the Takes on the disk untouched. Standard gating techniques can be applied to the audio, but with some powerful additions like programmable fades, pre-trigger time and floor attenuation.
It is possible to store your gate settings for easy recall. (Open/Save)



Noise Gate function parameters

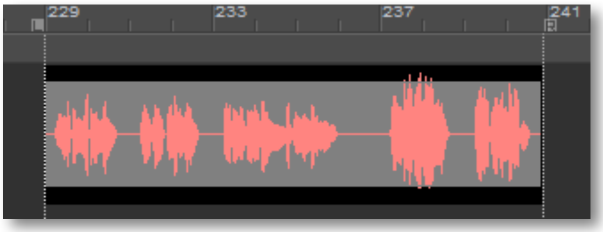
- **On Lvl** (-96..0dB): Threshold On Level
  Threshold level for opening the gate.
- **Off Lvl** (-96..0dB): Threshold Off Level
  Threshold level for closing the gate (i.e., the gate closes when the audio drops below this level).
- **Flr Lvl** (-96..0dB): Floor Level
  Level that is used for scaling 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 (i.e., allowing audio that starts just before the "On Lvl" is reached to be included).
- **Hold** (0.010..10sec): Hold Time
  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 (i.e. this sets the smallest Part length that will be generated).
- **Min Off** (0.000..10sec): Minimum "gate off" time
  "Gate off" region minimum length (i.e., 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.

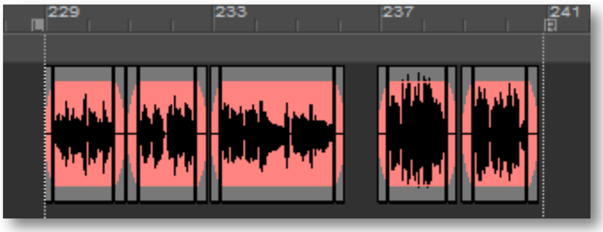
A good starting point for setting the thresholds is to set the “On Lvl” to 15dB below peak level and the “Off Lvl” to 15dB above the noise. The noise gate scans the audio in the Parts, taking into account the beginning volume of the Parts that exist before processing has taken place, i.e., you can pre-adjust 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 the following example a short vocal phrase was recorded:

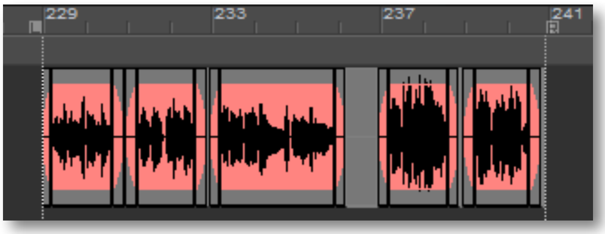


The noise gate processing was then performed, but the chosen parameters produced too many short Parts, removing some of the wanted sections of audio:

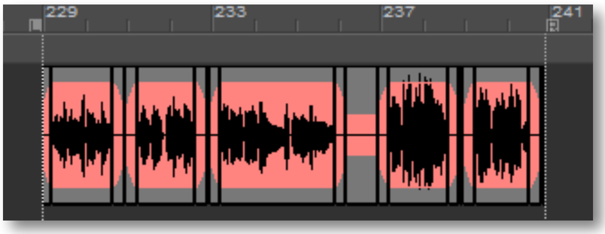
The noise gate parameters were modified to add a short fade in and fade out, we increased the “Hold”, “Min On”, and “Min Off” times and slightly lowered the thresholds. Reprocessing achieved the desired result, so that a gap in the vocal was removed:
In this example **Remove Parts where gate is off** 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, by using the Solo tool.



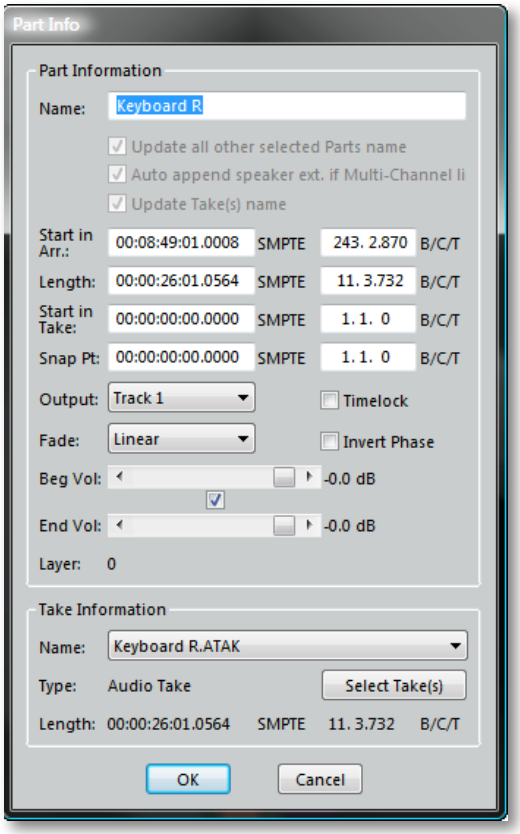
If **Floor Level Parts where gate is off** is selected, then instead of being removed in the “gate off” regions, the audio just becomes attenuated. For situations where completely removing the background ambience sounds unnatural, this can provide a controlled reduction of background levels.



Info tool [Ctrl]+[I]

This tool works on Audio and Automation Parts.

The Info tool is used to view or edit information about a Part or Take. It allows Parts and Takes to be named 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 dialog box will be displayed as shown:

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 the OK button 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" (e.g. L, C, R or S), depending on the setting of the **Multi-Channel speaker name extension** option under menu: Settings|Preferences|General.
- If the **Update Take(s) Name** box is checked, the name of the Takes referenced by the selected Part(s) will be updated when the OK button 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 Takes" button is dimmed if several Parts are selected and reference Takes located in different Folders (because such Takes 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" value(s) 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, even for 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, but 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 which type of automation data is shown in the Part (the Automation Curve Select tool, described in the "Mixer Automation" chapter, can also be used for that purpose).

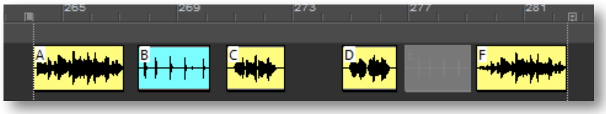
Consolidate Process tool [Ctrl]+[Shift]+[G]

This tool only works on audio Parts.

The Consolidate Process tool creates 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 that become muted are automatically deleted.

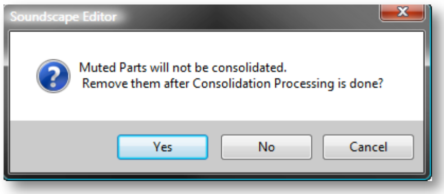
WARNING: The Takes generated by the Consolidation process only include the audio data that is actually used by the selected Parts. If the original Takes referenced by these Parts contain valuable audio data that is not used by the Parts in the current Arrangement, please note that this audio may be deleted after processing, depending on the option selected under menu: 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

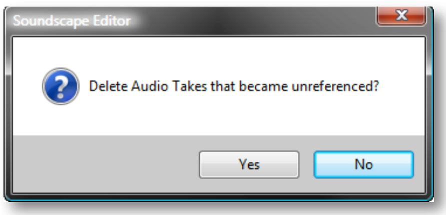
After clicking one of the selected Parts, because the selection includes a muted Part, the following dialog box is displayed. Clicking “Yes” or “No” triggers the processing, clicking “Cancel” aborts the operation.



Clicking “Yes” allows Part E to be removed from the Arrangement, resulting in only three parts. 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.

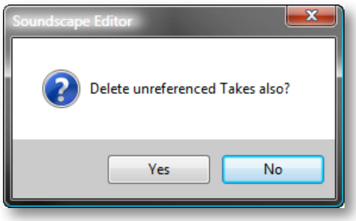
Clicking “No” allows the muted Part E to remain in the Arrangement.

A further dialog box may be displayed, depending on the option selected under menu: Settings|Preferences|Arrangement|Audio Processing tools behaviour|Audio Takes that become unused after processing (the available options are described in the Preferences section of the Settings Menu chapter).



Delete tool [Ctrl]+[D]

This tool works on Audio and Automation Parts.



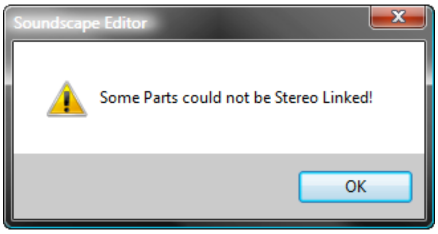
Click on a Part or group of selected Parts to delete it. If, when the Parts you have selected are deleted, Takes are left on the record/process folder which have become unreferenced in the current Arrangement, a dialog box will be displayed.
If you no longer require these Takes, and you are sure that they are not used by any other Arrangement, click “Yes”. The Takes will then be deleted from the disk. Click “No” if you want to save the Takes, or if you are not sure whether other Arrangements use them.

NOTE: When a Part or group of Parts has been selected, it is also possible to press the [Delete] key instead of using the Delete tool.

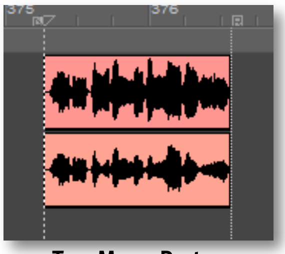
Stereo Link tool [Ctrl]+[L]

This tool only works on audio Parts.

The Stereo Link tool allows two Parts to be linked for all editing operations, including soloing. To stereo link the 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, reference Takes of identical word length, and are both assigned or both unassigned (to a track — even if either or both are muted). If this is not the case, a dialog box appears.



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



Two Mono Parts

Stereo linked Part

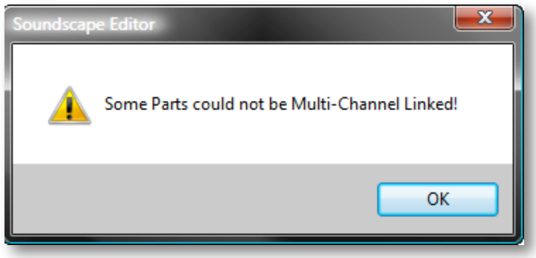
The same tool can be used to undo the stereo link between two Parts. Just click either of the linked Parts and they become independent again.

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.

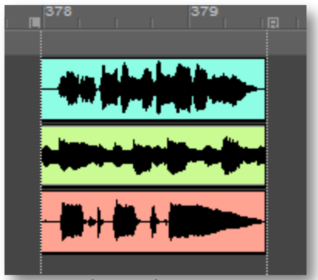
Multi-Channel Link Part tool [Ctrl]+[4]

This tool only works on audio Parts.

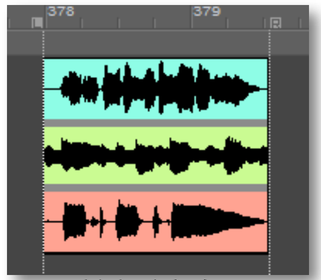
The Multi-Channel Link Part tool allows multiple Parts to be linked for all editing operations, including soloing. To multi-channel link the Parts, select them then click either of them. Multi-channel linking can only take place if the Parts are on adjacent virtual tracks, have identical lengths and start positions in the Arrangement, reference Takes of identical word length, and are all assigned or all unassigned (to a track - even if some of the Parts are muted). If this is not the case, a dialog box appears .



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 colors. Using the Track Assign tool on multi-channel linked Parts will assign tracks in series (i.e., 1-2-3, 3-4-5, etc.).



Independent Parts



Multiple Linked Parts

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.

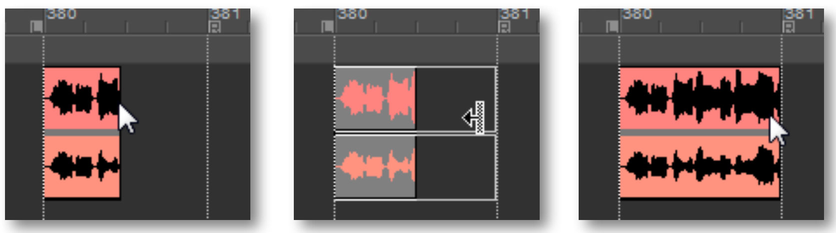
NOTE: If you use the Multi-Channel Link tool on a group of Parts, the entire group will take on the same link status as the Part that you click on, regardless of each Part's prior status.

Trim Tool (Exclude Adjacent Parts Unless Selected) [Ctrl]+[T]

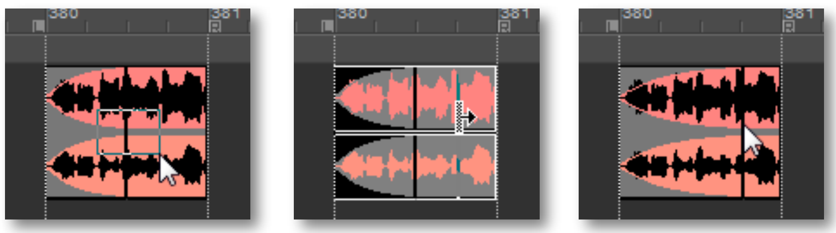
This tool works on Audio and Automation Parts.

The Trim Tool (Exclude Adjacent Parts unless selected) 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, and this allows 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, for instance, when you have cut a Part in two with a fade into the second section. The Trim (Exclude Adjacent Parts unless selected) 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) [Ctrl]+[Alt]+[T]

This tool works on Audio and Automation Parts.

The Trim Tool (Include Adjacent Parts) 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, i.e., 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

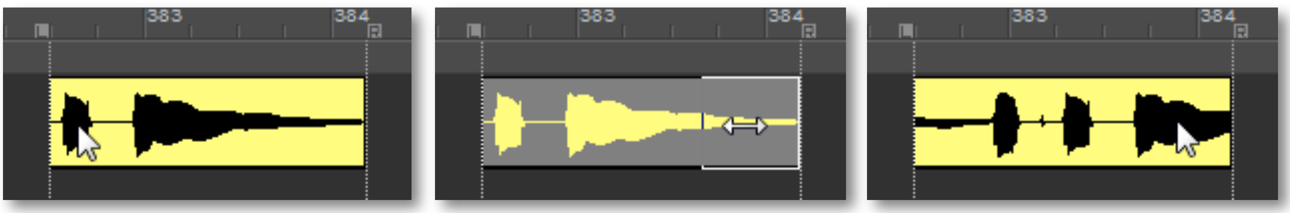
A Part or a group of selected Parts can be trimmed in increments defined by the current snap value, if the Snap function is active, by clicking and holding it with the tool and using the [Left Arrow] and [Right Arrow] keys. If the Snap function is inactive increments of one sample will be used.
The length of a Part can only be extended if it is shorter than the referenced Take, i.e., if there is still available audio data on the disk.

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

 **Slip tool** [Ctrl]+[J]

This tool works on Audio and Automation Parts.

The Slip tool 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 the disk is longer than the Part in the Arrangement, and so there is excess data on at least one end.



Select the slip tool, click on the audio (waveform) that you wish to move and drag left or right while holding the mouse button. The pointer shape will change to a double-pointed arrow and an outline which shows how much movement is being made will be drawn, as shown in the second picture above. If the Snap function 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.

The data within a Part or group of selected Parts can be slipped in increments defined by the current snap value, if the Snap function is active, 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 the Snap function is inactive, increments of one sample will be used.

NOTE: Once the data within a Part has been moved using the slip tool, this 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 should join. If you need the data from two such Parts to be included in one single new Part, please use the Consolidate tool.

Phase Invert tool [Ctrl]+[P]

This tool only works on audio Parts.

The Phase Invert tool allows individual Parts to be 180° inverted in real-time (i.e., this is a playback parameter and there is no need for a new Take to be generated on the disk). The Phase Invert status of a Part is reported in the Part Info window and can also be changed there:

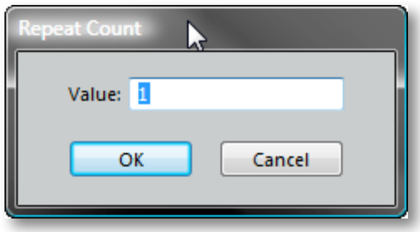
Phase inversion may be necessary if equipment in the record or playback audio path is also inverting. This can be corrected easily using the Phase Invert tool on individual Parts or entire tracks.

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 has the effect of implementing a simple comb filter, which will cancel some frequencies and enhance others, depending upon the delay.

Repeat tool [Ctrl]+[R]

This tool works on Audio and Automation Parts.

The Repeat tool allows a Part or a group of selected Parts to be repeated any number of times in the Arrangement. If the snap setting 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 the snap setting is not active, (or if the snap setting is active but 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 that must be repeated calls up the following dialog box, where the required number of repeats must be entered before clicking "OK":

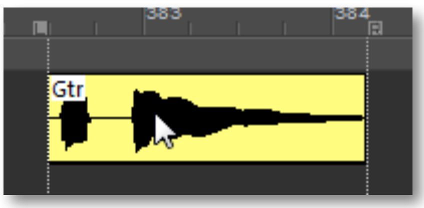


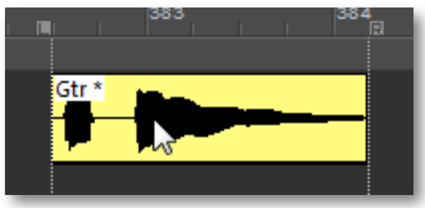
NOTE: If the number of repeats entered causes an overlap with other Parts in the Arrangement and overlapping Parts are not allowed under menu: Settings|Preferences|Arrangement|Overlapping Parts (for new edits), a warning message will be displayed. In this case, the maximum number of repeats possible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

Timelock tool [Ctrl]+[Shift]+[L]

This tool works on Audio and Automation Parts.

The Timelock tool 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.





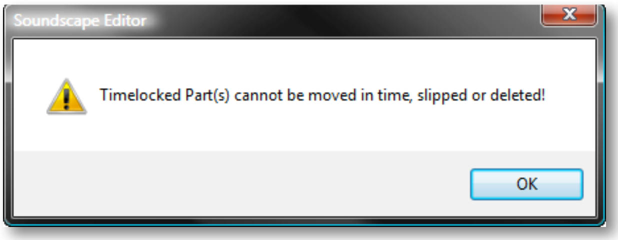
A timelocked Part has a "*" character appended to its name as shown on the right screenshot.

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, but 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 following message 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 track(s) which do not contain timelocked Parts.

 **Automation Curve Select tool** [Ctrl]+[Shift]+[A]

This tool only works on Automation Parts.

The Automation Curve Select tool is used to select which automation curve is displayed for an automation Part that contains several automation curves.

It is described in detail in the “Mixer Automation” chapter of this manual.

 **Automation Event Editing tool** [Ctrl]+[Shift]+[E]

This tool only works on Automation Parts.

The Automation Event Editing tool is used to create, delete or edit automation events within existing automation Parts.

It is described in detail in the “Mixer Automation” chapter of this manual.

 **Automation Events Thinning tool** [Ctrl]+[Shift]+[F]

This tool only works on Automation Parts.

It is used to “thin” automation data by a user-selectable factor.

It is described in detail in the “Mixer Automation” section of this manual.

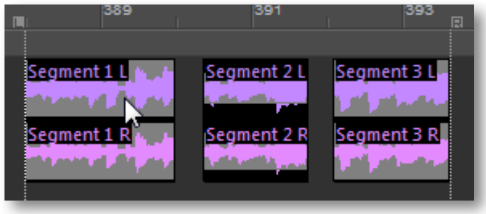
 **Crossfade tool** [Ctrl]+[Shift]+[X]

This tool only works on audio Parts.

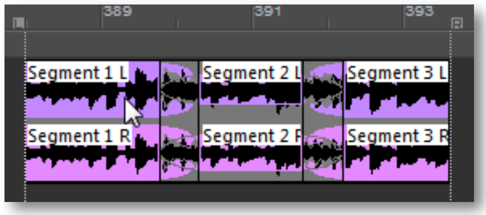
The Crossfade tool allows the crossfading of any number of selected successive audio Parts on the same virtual track providing that these Parts overlap. If the Parts do not overlap but reference Takes that have enough excess audio at their respective end of the projected crossfade area, they will be 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.

Also, 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).

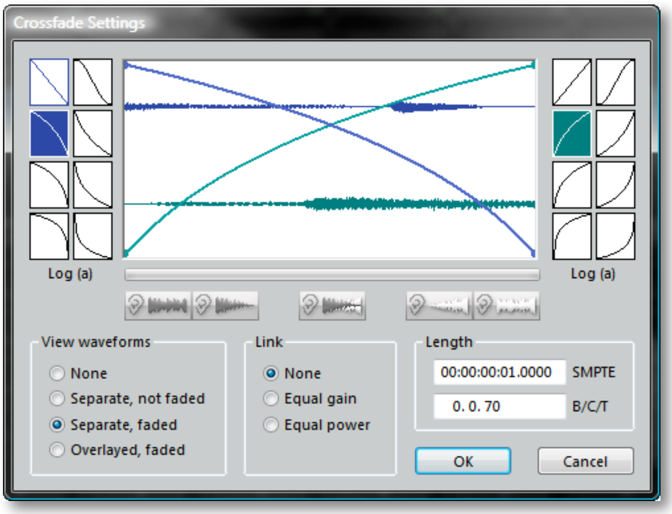
In the following example, three consecutive pairs of stereo linked Parts with no overlaps are selected, then the Crossfade tool is used.



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 vertical 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 will be applied for each crossfade you subsequently create. A similar window can be accessed by clicking with the tool in any existing crossfade area 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 first Part ("Fade Out Part") and second Part ("Fade In Part") are shown respectively in blue and green. The eight possible curve shapes are shown on either side of the graphical display. The currently selected curve has a blue ("Fade Out Part") or green ("Fade In Part") background, and the other selectable curve shapes have a white background. If "Equal power" is selected in the "Link" section of the window the 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. Therefore, if the "Fade Out Part" has a volume of -4dB at the crossfade start point, the fade out will be from -4dB to -96dB, and if the "Fade In Part" has a volume of -3dB at the crossfade end point, the fade in will be from -96dB to -3dB.

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

 The "Fade Out Part" on its own, fade NOT applied.

 The "Fade Out Part" on its own, fade applied.

 Both Parts with the crossfade applied.

 The "Fade In Part" on its own, fade applied.

 The "Fade In Part" 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 the way the audio is shown on screen and this does not affect playback at all.

Link

- If you select **None**, any fade curve shape can be chosen independently for the “Fade Out Part” and “Fade In Part”.
- If you select **Equal gain** you can choose any fade curve shape for either the “Fade Out Part” or “Fade In Part”, and the fade curve shape for the other one will be selected automatically so that it “mirrors” the chosen one. For example, if the “Fade Out Part’s” gain drops slowly, the “Fade In Part’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 Part” or “Fade In Part”, 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

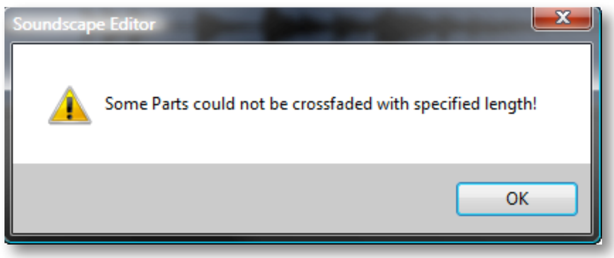
When a crossfade is created between non overlapping Parts, its length can be specified before using the tool, in the Crossfade Settings window accessed by double-clicking on the Crossfade tool icon. The required crossfade length can be entered either in SMPTE + sample extension or in B/C/T (bars/counts/ticks) 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 [Ctrl]+[2]

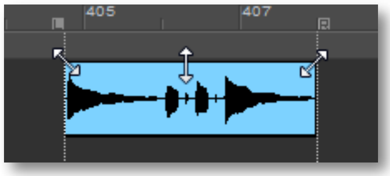
This tool works on Audio and Automation Parts.

By default this single, fully customisable timesaving 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 placement 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 possible pointer shapes are all very evocative, and often identical to the ones shown when using the dedicated tools, which makes using the Context Sensitive Edit tool very intuitive. 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.

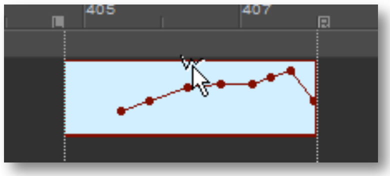
NOTE: Selecting the Context Sensitive Edit tool will automatically cancel all previous tool selections. This is normal and due to the fact that the Context Sensitive Edit tool makes use of both mouse buttons and the [Alt] key.

Using the left mouse button

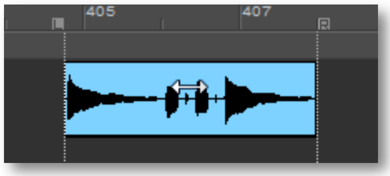
If the pointer is in the upper section of an audio Part, the Context Sensitive Edit tool operates as the Volume Trim tool.



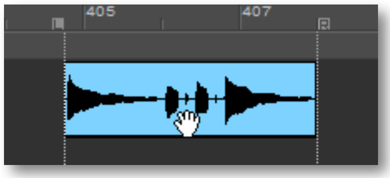
If the pointer is in the upper section of an automation Part, the Context Sensitive Edit tool operates as the Automation Curve Select tool.



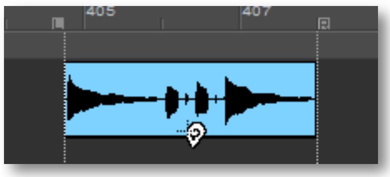
If the pointer is in the upper middle section of a Part, the Context Sensitive Edit tool operates as the Slip tool.



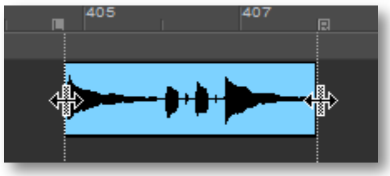
If the pointer is in the lower middle section of a Part, the Context Sensitive Edit tool operates as the Move tool:



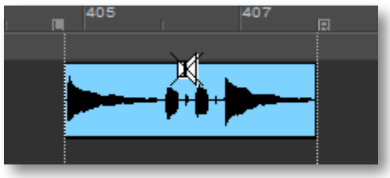
If the pointer is in the lower section of a Part, the Context Sensitive Edit tool operates as the Solo tool:



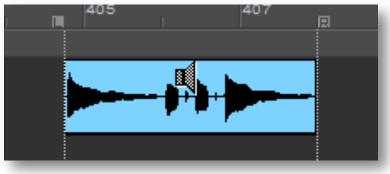
If the pointer is near the beginning or end boundary of a Part, the Context Sensitive Edit tool operates as the Trim (Include Adjacent Parts) tool:



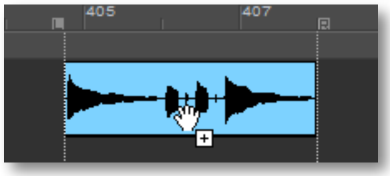
With the [Alt] key held down, if the pointer is in the upper section of a Part, the Context Sensitive Edit tool operates as the Mute tool:



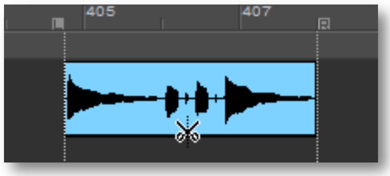
With the [Alt] key held down, if the pointer is in the upper middle section of a Part, the Context Sensitive Edit tool operates as the Track Assign tool:



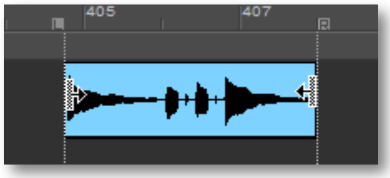
With the [Alt] key held down, if the pointer is in the lower middle section of a Part, the Context Sensitive Edit tool operates as the Copy Tool:



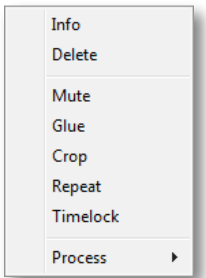
With the [Alt] key held down, if the pointer is in the lower section of a Part, the Context Sensitive Edit tool operates as the Cut tool:



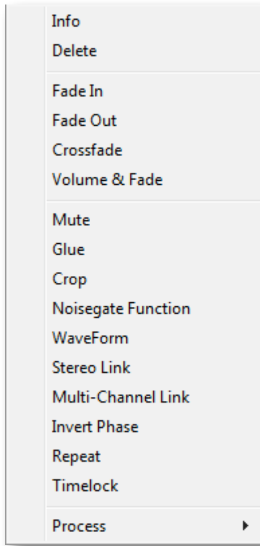
With the [Alt] key held down, if the pointer is on the edge at the beginning or end of a Part, the Context Sensitive Edit tool operates as the Trim tool:



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):

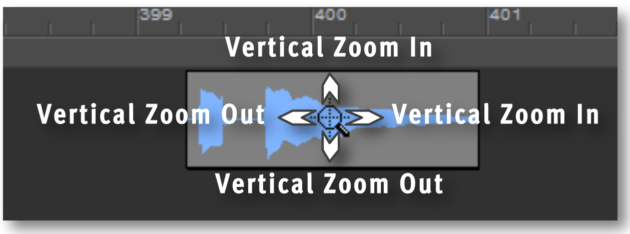


Automation Part

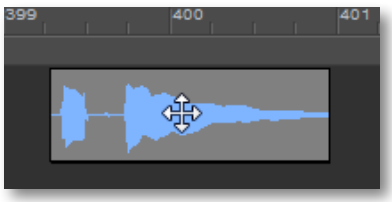


Audio Part

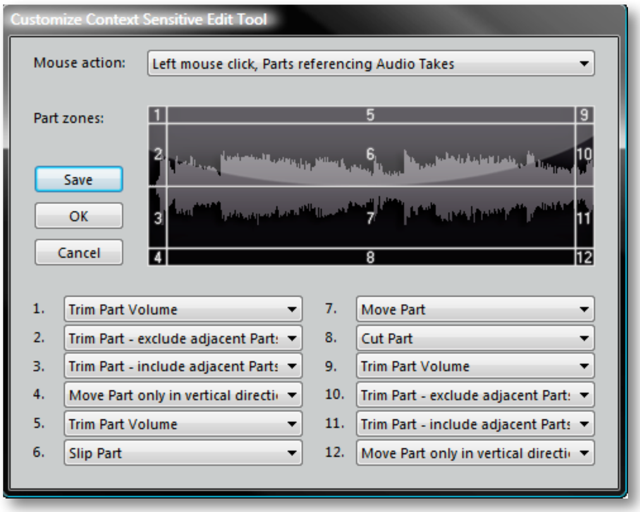
Dragging a Part with the right mouse button adjusts the horizontal or vertical zoom level according to the dragging direction. The point where the Current Locator is located when the dragging movement starts is the “center” of the zoom (i.e., it does not move and therefore remains visible). If you are 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



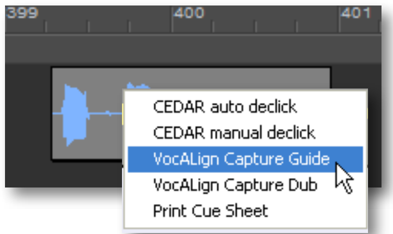
Once you have got to grips with the Context Sensitive Edit tool, you may want to customise its operation to suit your preferences. This can be done easily: double-clicking the Context Sensitive Edit tool icon in the Toolbar opens a window where you can select any editing function for each one of twelve 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 [Ctrl]+[Alt]+[E]

This tool only works on audio Parts.

The XPro tool is 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 the SSL Soundscape V6 (its operation is described in the "XPro Cue Sheet Printing" chapter).

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.

Mix Window

Soundscape V6'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.

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 separate from the Project, making it possible to load different Mixes for a Project.

Also unlike other DSP based DAW's it takes a lot less than 1 sec 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 (e.g. 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 V6 is also capable of working as a Mixing front end for any other ASIO DAW Software.
For further Details please read **Chapter 7 of the SSL Mixer V6 User Guide** (in Soundscape V6: menu Help|Mixer Manual)

Create, Load and Save a MIX

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 V6.

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 V6 mixer files is: **C:\Soundscape\Mix\Soundscape**

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

In order to Save your changes click menu **File|Save Mix as...**

Mixer building Blocks

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.

Mixer Strips and DSP Processing Power

It is possible to "DSP mute" mixer strips or mixer elements using the Mute tool (i.e. "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 with duplicated strips having 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%, etc.
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 (i.e., 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

DSP Resources and DSP Auto Routing

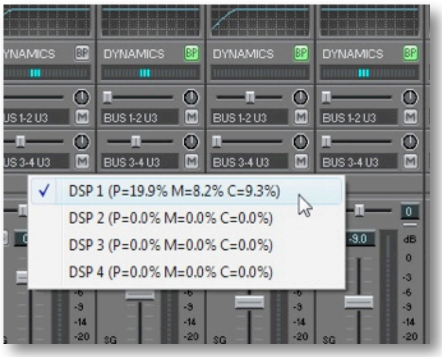
Assigning a mixer strip to a specific DSP

NOTE: This section assumes that Mixer Edit mode is selected (press [E] on the keyboard to switch between Mixer 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, shows 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.



Mixer auto routing

Auto routing can be enabled or disabled by ticking or unticking the "Auto route" item under menu: Settings|Preferences|Mixer.

Mixer auto routing enabled

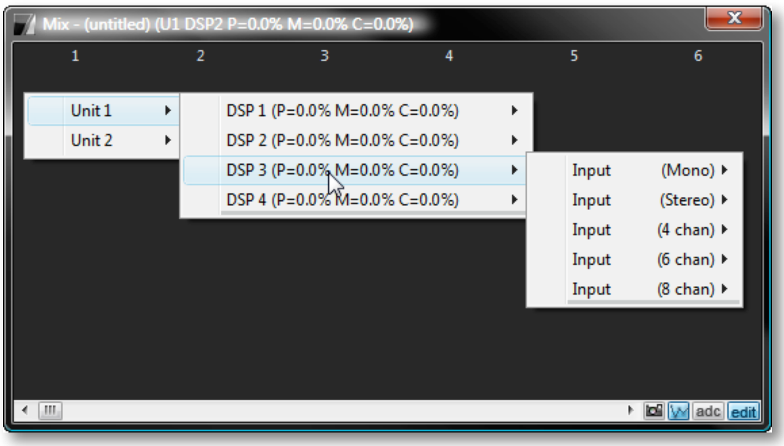
If the "Auto route" item under menu: Settings|Preferences|Mixer is **ticked** (auto routing enabled), the DSP resources are allocated automatically.

Mixer auto routing disabled

If the "Auto route" item under menu: Settings|Preferences|Mixer is **unticked** (auto routing disabled), the DSP resources need to be allocated manually when building a Mixer.

When the **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).



Basic Mixer Strip Structure

This diagram shows signal routing through a stereo input mixer strip.

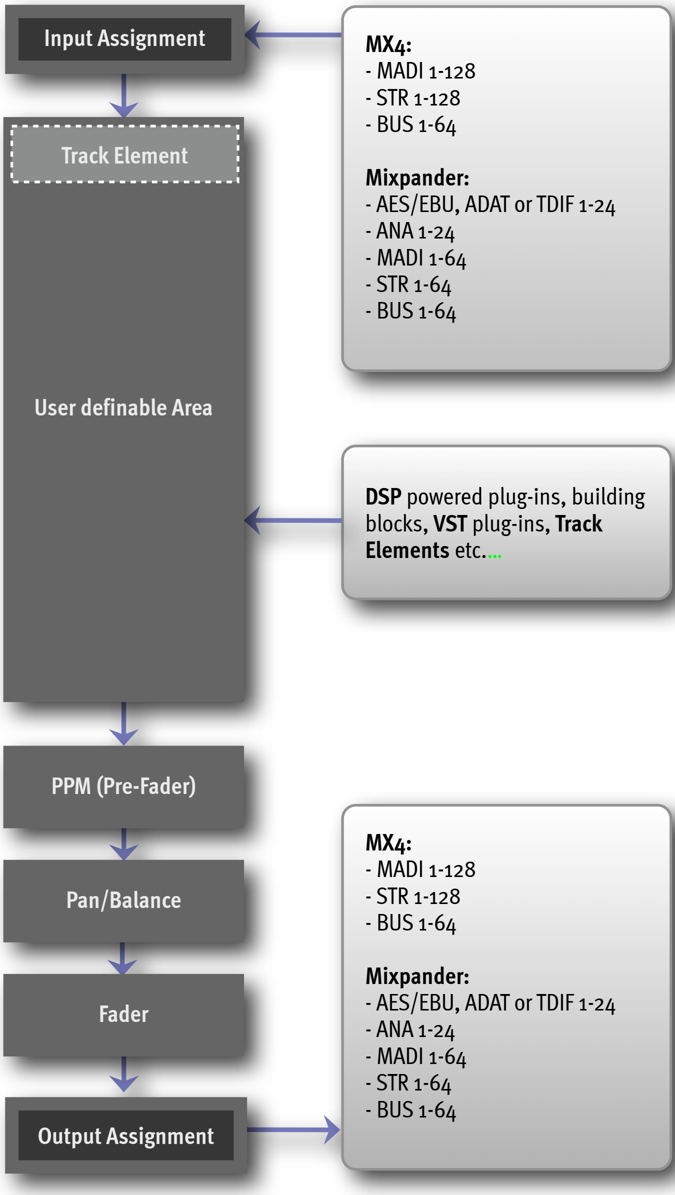
The signal can be fed via any of the external inputs, it can be internally sent from another mixer strip via one of the "buses" or it can be received from a Windows application via a streaming input.

The Track Elements represent the "Tape" Tracks, where the Audio can be routed to/from the 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, the level can be adjusted and monitored, it can be delayed, sent to another mixer strip or output and processed by one of the available "plug-ins".

The fixed peak meters are pre-fader so they will show a signal even if the fader is all the way 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, i.e. 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.



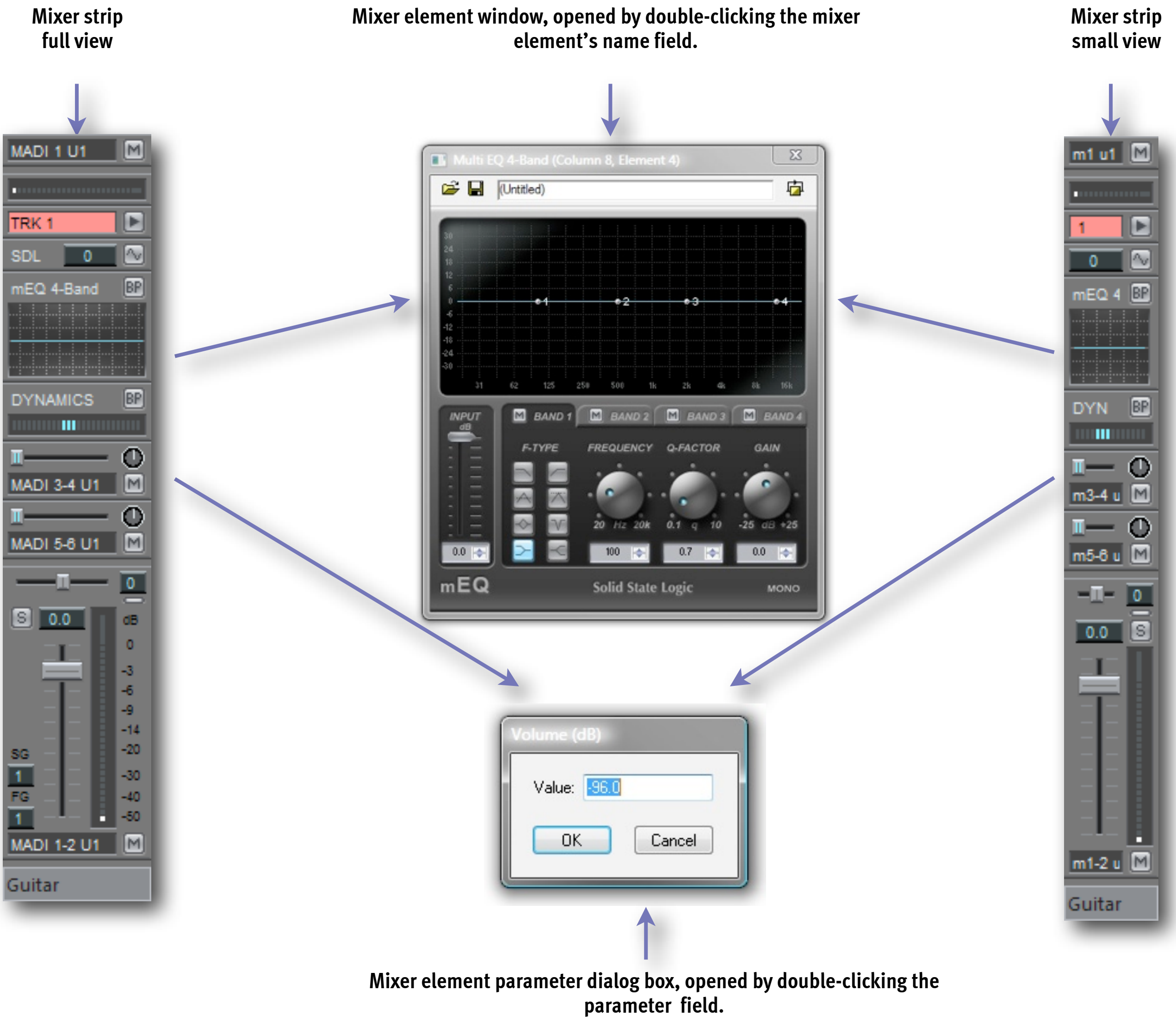
Mixer Views

The SSL Mixer can be displayed in one of two modes: full column size (wide) or small column size (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 column size mode up to 11 mixer columns can be displayed across the screen. In small column size mode this number is increased to 17, but less detail is visible.

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 dialog box where the value for that particular parameter can be edited with text input on the keyboard.

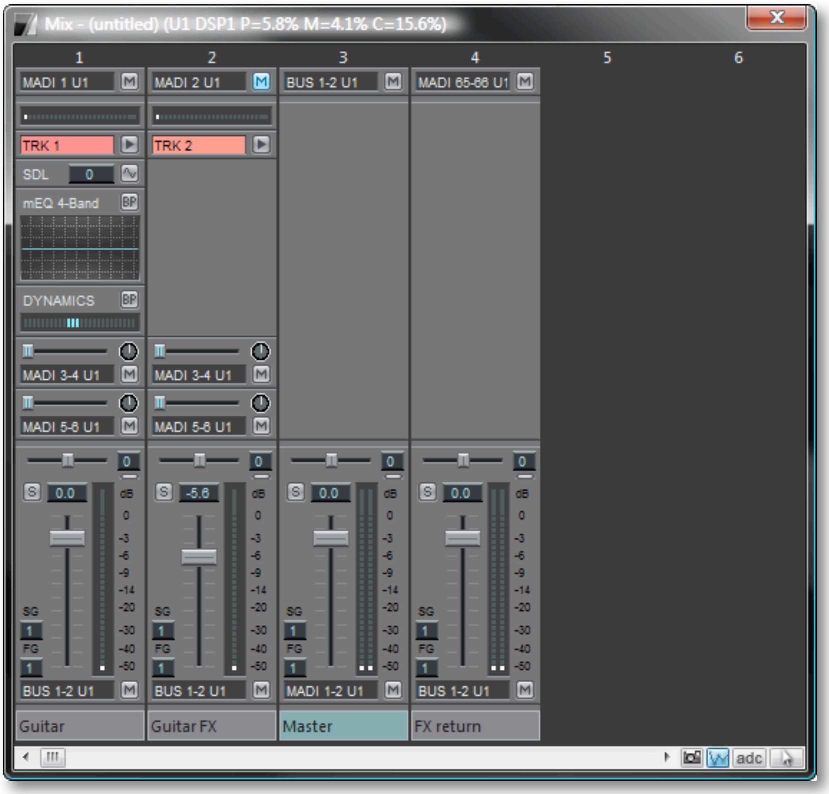


NOTE: Once a Mixer has been loaded, it remains active even if the window is closed.

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 when necessary, i.e., whenever one or more mixer strips cannot be entirely viewed in the mixer window. This is necessary because while the Mixer window can be resized, **there is no limit to the number of mixer elements** that can be inserted **inside the user definable area**, and therefore there is no limit to the vertical size of a mixer strip.



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.

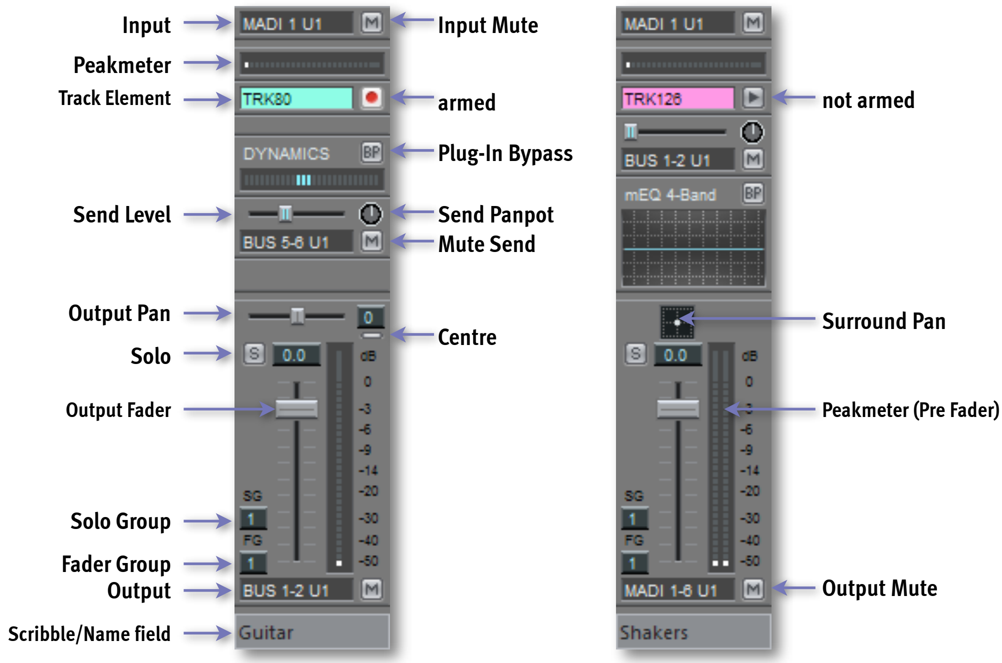
Notice that only the first mixer strip has an EQ and Compressor, while both have a send element.
This is an example of putting power where it is needed.
The restrictions of some hardware mixing desks, and of inflexible, predefined software mixers that do not reach SSL's workflow and flexibility standards have been relegated into the past and are now available to directly inside the SSL DAW.

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!

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 LRC-LsRs (3.1 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 dialog 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 could can contain several output elements, each assigned to different fader and solo groups.

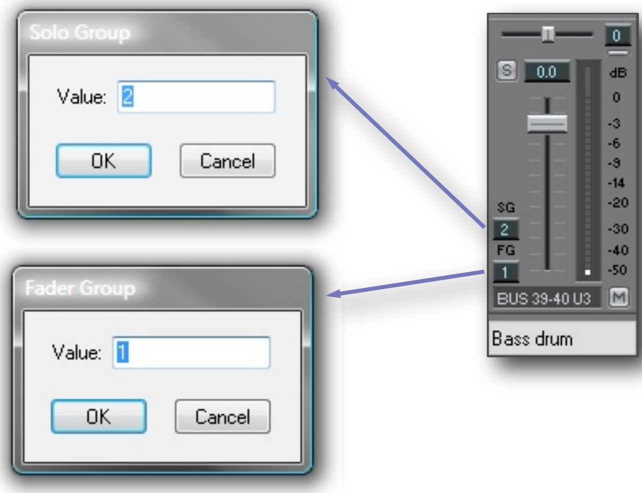
The **fader group and solo group assignments** can be selected in several ways:

- 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 as well to speed up the process even more.
- 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 dialog 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

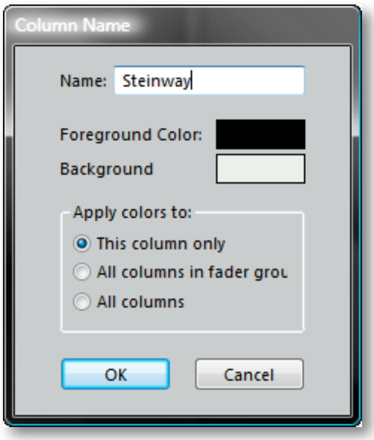
- **Left-clicking** any inactive (grey) solo button solos the corresponding output and silences all other outputs that have the **same solo group assignment.**
- **Any previously soloed output inside the same solo group** is dropped out of solo mode and silenced, unless the [Ctrl] key is used as described 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 all soloed outputs turn red, and the solo buttons of the silenced outputs display a red "S" on 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 Mixer 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, that 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.

Input/Output Identification

Depending on the SSL hardware that is used with the SSL Mixer V6, 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 with 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.

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 SSL Soundscape V6 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, e.g. 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 Editing Tools

The Soundscape Toolbar contains seven tools to create, edit and fully configure the SSL Mixer's structure for each channel.
Although simple to use, the tools give you the power to design almost any structure you can think of.

NOTE: The Tools only work when you are in **Mixer Edit Mode**.

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. 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 other 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.

Edit Mode Tools

 **Create tool:** Create new Strips in empty columns or new mixer elements in the User definable area of the Strip

 **I/O Assign tool:** Change I/O's for Inputs/Outputs and Sends, assign Strip to another DSP

 **Mute tool:** DSP mute mixer elements or complete mixer columns

 **Move tool:** Move mixer elements inside Strip or mixer strips to another column (additionally Move vertical and Move to Loc tools can be used)

 **Copy tool:** Copy mixer elements inside Strip or mixer strips to another column (additionally Copy vertical and Copy to Loc tools can be used)

 **Delete tool:** Delete mixer elements or mixer columns

 **Info tool:** used to display information about mixer elements or mixer column

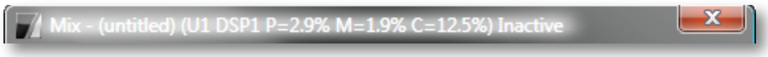
Each Tool's function is explained in detail in Chapter 9. Menu Reference, Section Edit Menu.

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.

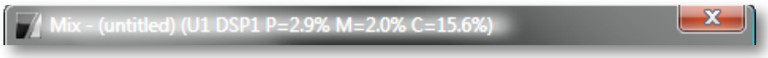
Mixer Inactive Build

For each change that is made in Edit Mode the DSP code needs to be updated, potentially all of the DSP code has 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 or in other words: how big your mixer is, this could take from a couple of milliseconds to approx. 1 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 only once.
If native mixer elements such as VST plug-ins are used, native processing is also stopped while the Mixer is being edited.

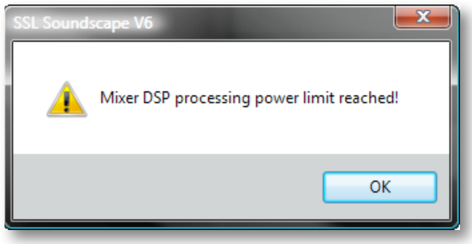
To use Mixer Inactive Build, hold the [Shift] key down while performing the required edits. The title bar will display "Inactive" after the "P", “M“ and "C" values, and the changes you make will be reflected instantly on screen.



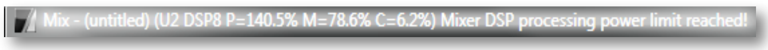
The [Shift] key can be released before the last edit is carried out in order to reactivate the edited Mixer.
Alternatively you can double click on the Caption bar to reactivate the Mixer.

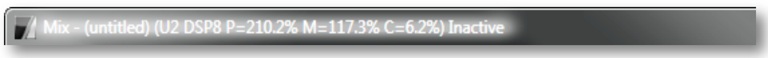


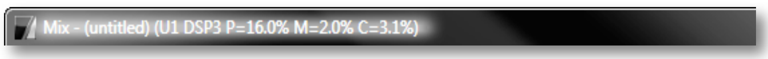
DSP power limit
The "P" value reported in the Mixer window’s title bar is the percentage of the available DSP clock cycles required to run the Mixer at the selected Sample Rate (taking the Varispeed setting into account). The "M" value is the percentage of memory resources required to run the Mixer. 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.

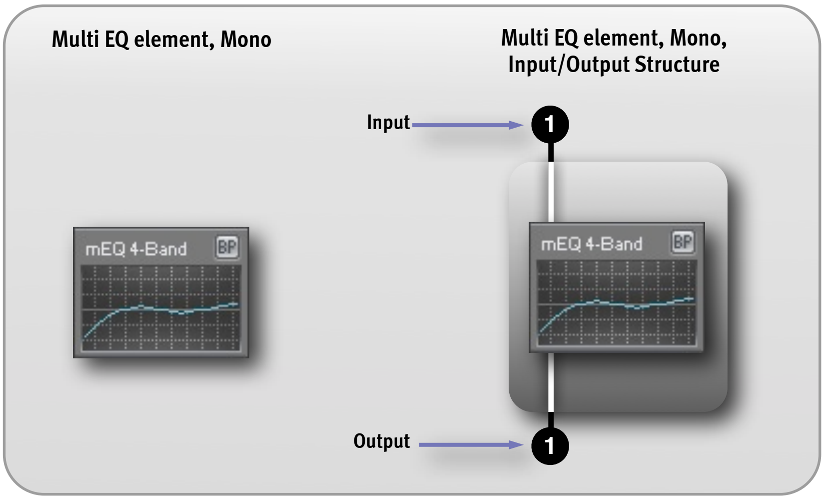
However, the P=xx% value is not always completely accurate, because some plug-in elements may report inaccurate processing cycles. This means that depending on the elements used, some Mixers can't reach or may exceed the P=100% value without any warning box appearing.
Also the Mixer DSP measures the 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

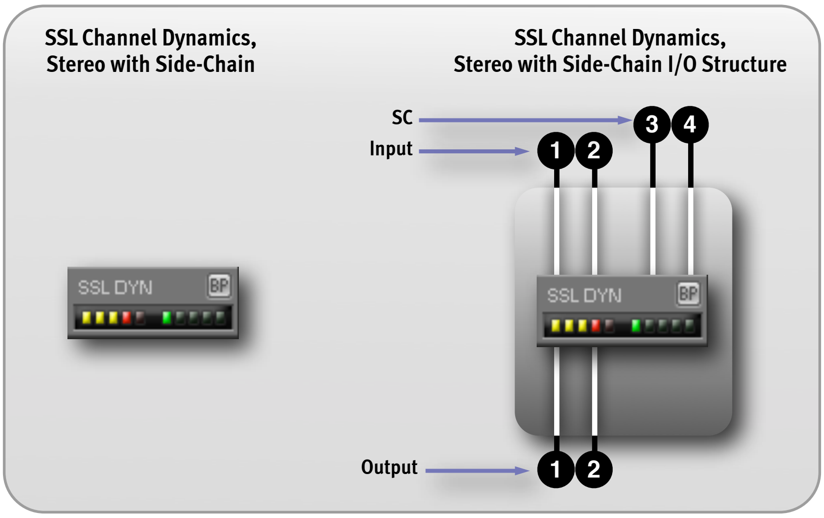
Each mixer element has at least one input and one output, and some elements have many more. These individual inputs and outputs come in three categories: **"fixed", "assignable", and "track element I/O",** which is a mixture of both.

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 black circles incremented from left to right. The examples on the right side of the page show a mono multi EQ mixer element, as it appears within a mixer strip, and with its single input and single output represented.



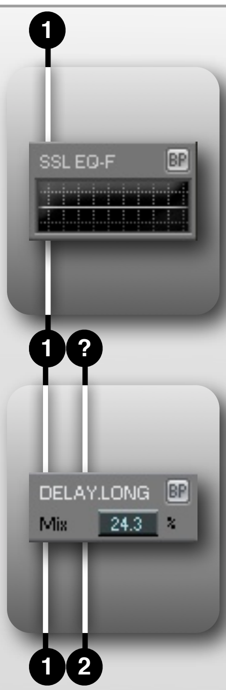
...and a stereo SSL Dynamics element with side-chain, as it appears within a mixer strip, and with its four inputs and two outputs represented.



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:

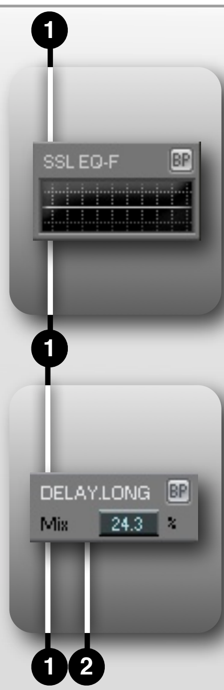
Incorrect configuration:

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)

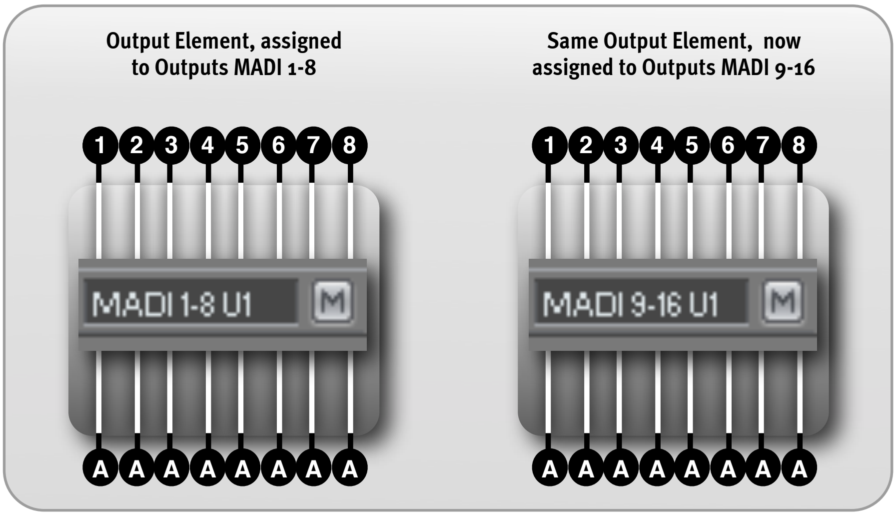


It is important to take this into account when building a Mixer, since the two configurations shown above would look the same within a mixer strip. If you need to check the input/output configuration of an existing mixer element, click on it with the Info tool in Mixer Edit mode. A window containing 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", and this means 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.

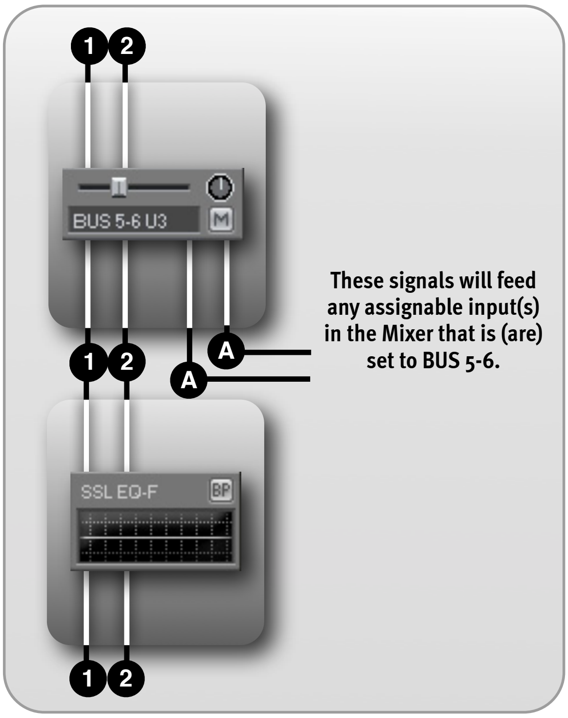


The example above shows an 8 in/8 out output element. Initially, the element's outputs are assigned to outputs 1-8. The I/O Assign tool is then used to change the element's output assignment to outputs 9-16.

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**, (i.e., unaffected by these settings and regardless of mute status).

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



Track Elements

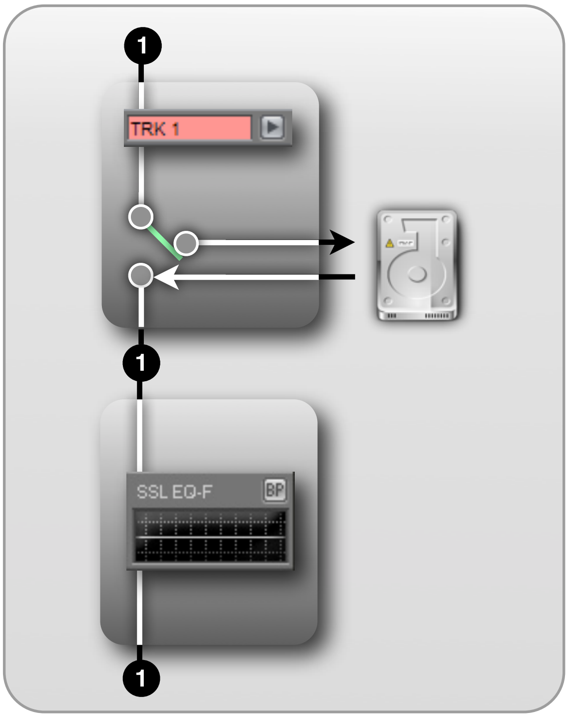
The Track Elements are a mixture of fixed and assignable I/O's.
Track Elements always connect to Audio Tracks in the Arrange window (fixed), however the Track they are connected to is assignable.

In Play Mode (Track not armed) the Input of the Track (1) is available to be mixed with the Audio coming from Disk (Double Click on Track Element and check **Mix Input Always**) at the Tracks Output.

In Record Mode (Track is armed) the Input of the Track is always routed to the Disk and from there back to the Tracks Output (independent of the selected **Mix Input Always** status).

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.



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 column Input/Output configurations

When you select the Create tool and click in an empty mixer slot in "Mixer 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 (i.e., 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.

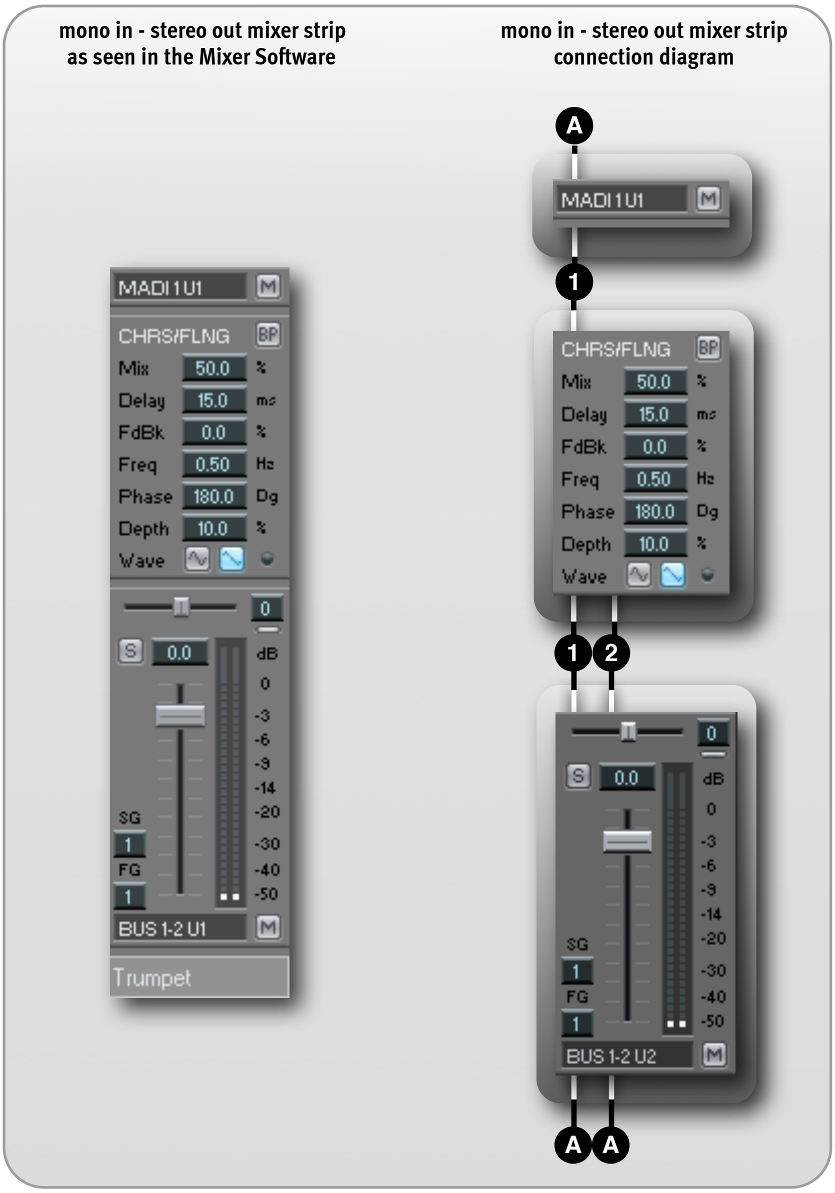
For mono in - stereo out, the input is mono, but the input signal needs to reach both inputs of a stereo output element. This mixer column configuration should only be created in order to use a stereo process or effect on a mono source (e.g., a mono to stereo chorus or reverb).

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 side, 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.

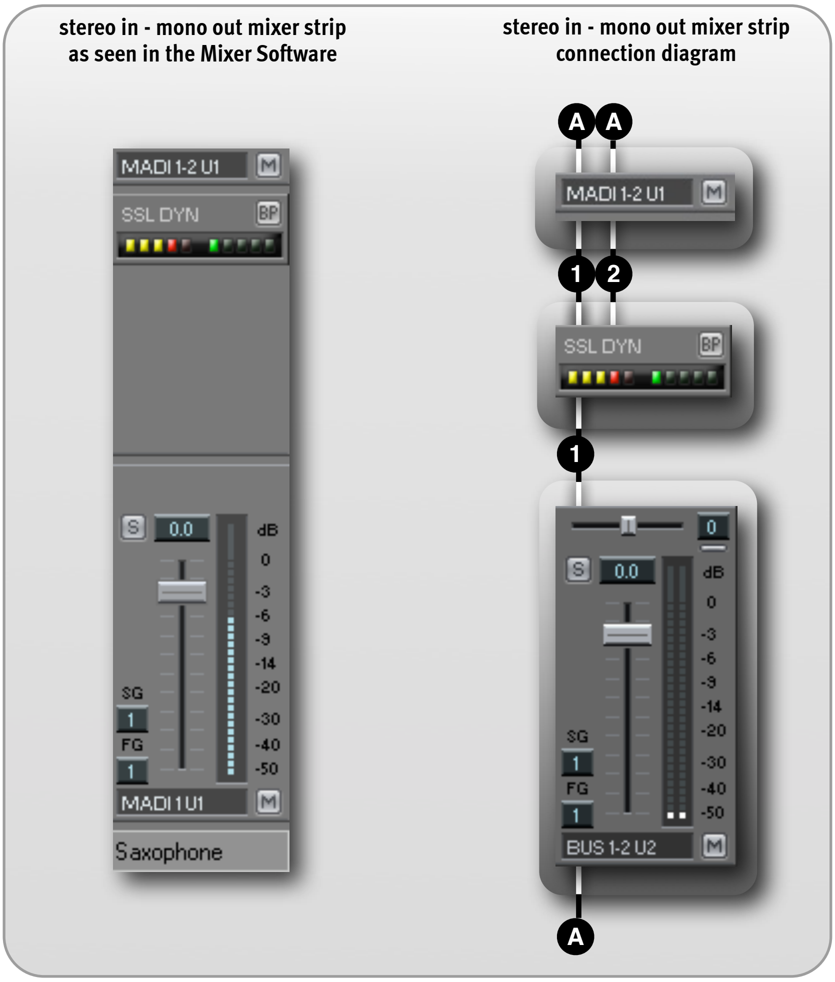


For **stereo in - mono out**, only the left side of the stereo input will be connected to the mixer column's output.

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.

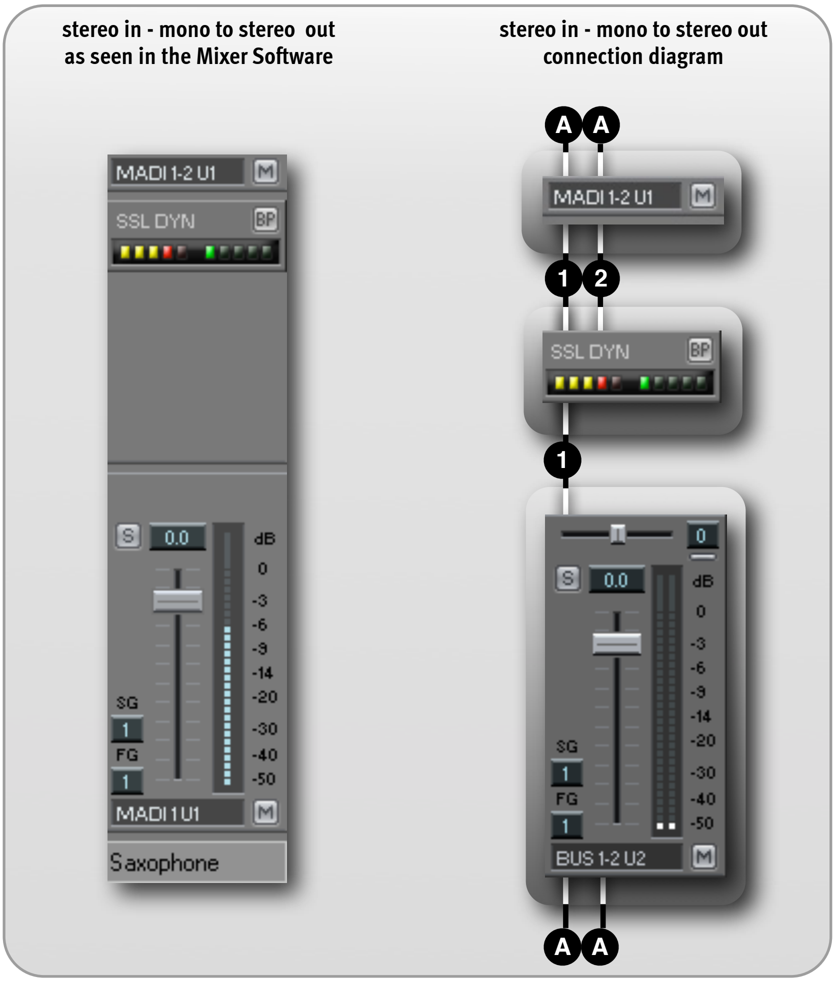
Only the main signal will reach the compressor's single output.



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 Balance and Panning

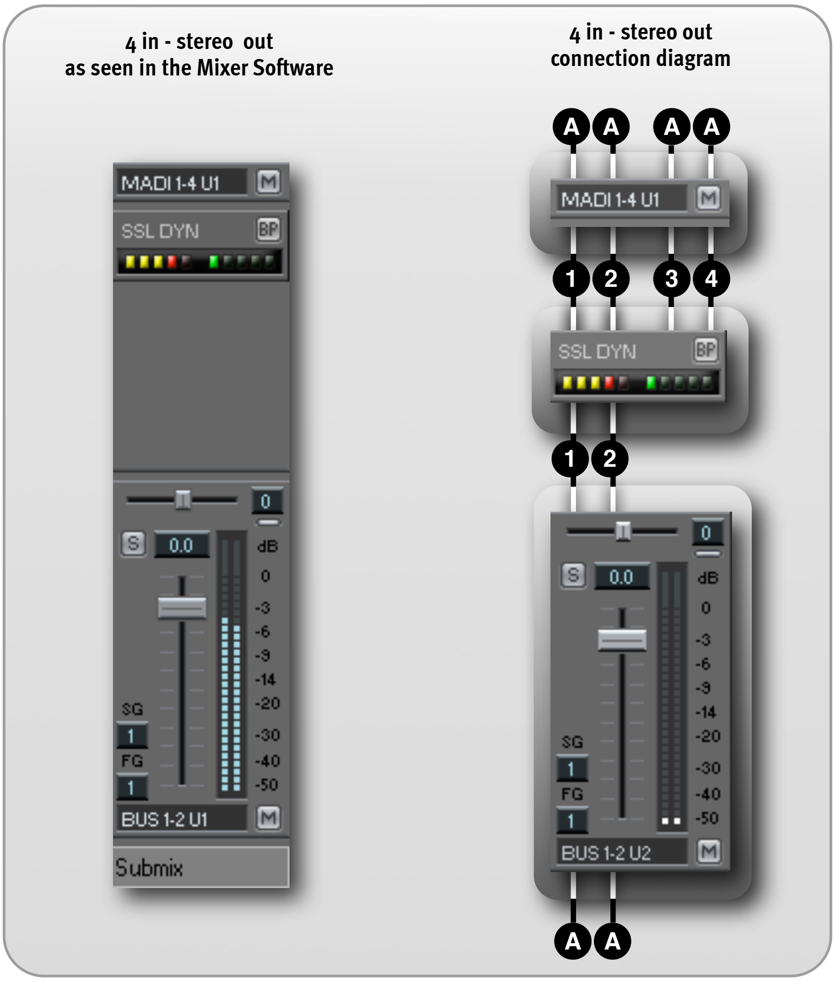
Stereo and mono to stereo outputs can be selected with **Equal Power Panning (EPP menu options)**.
In this case, when the balance or pan control is moved to one side, the level for that side is **gradually increased by up to 3dB** to compensate for the overall loss of power from the other side.

4 in - stereo out mixer columns

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.



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.**

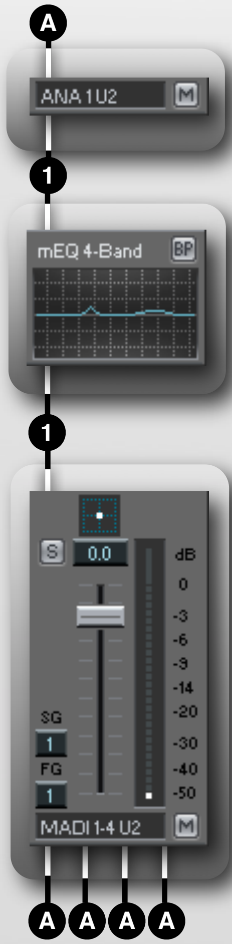
A **stereo in - stereo to LRCS out** configuration is similar to the mono in- mono to LRCS out configuration, except that the signal path through the mixer column is stereo, allowing stereo processes or effects to be applied to the signal.

Routing in this example: In this example the output routing of the LRCS signal is:

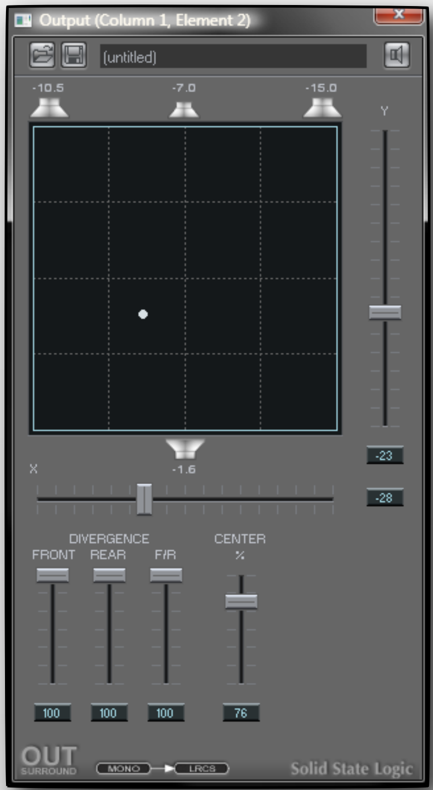
- MADI 1: L
- MADI 2: R
- MADI 3: C
- MADI 4: S

mono in - mono to LRCS strip as seen in the Mixer Software

mono in - mono to LRCS 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

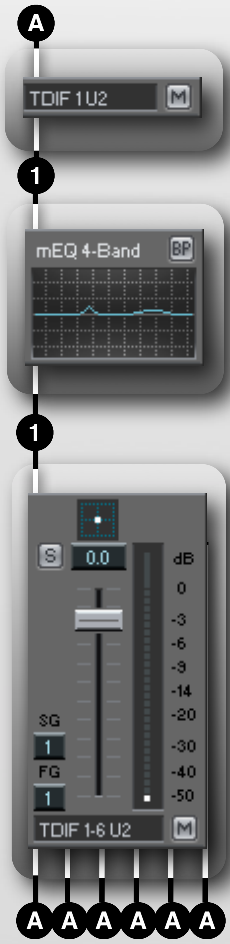
The **stereo in - stereo to 5.1 out** configuration is similar to the mono in- mono to 5.1 out configuration, except that the signal path through the mixer column is stereo, allowing **stereo processes** or effects to be applied to the signal.

Routing in this example:

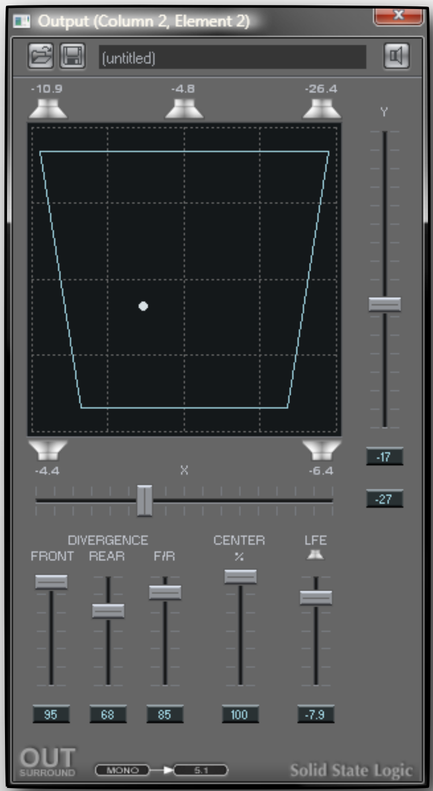
- MADI 1: L
- MADI 2: R
- MADI 3: C
- MADI 4: LFE
- MADI 5: L S
- MADI 6: R S

mono in - mono to 5.1 strip
as seen in the Mixer Software

mono in - mono to 5.1 strip
connection diagram



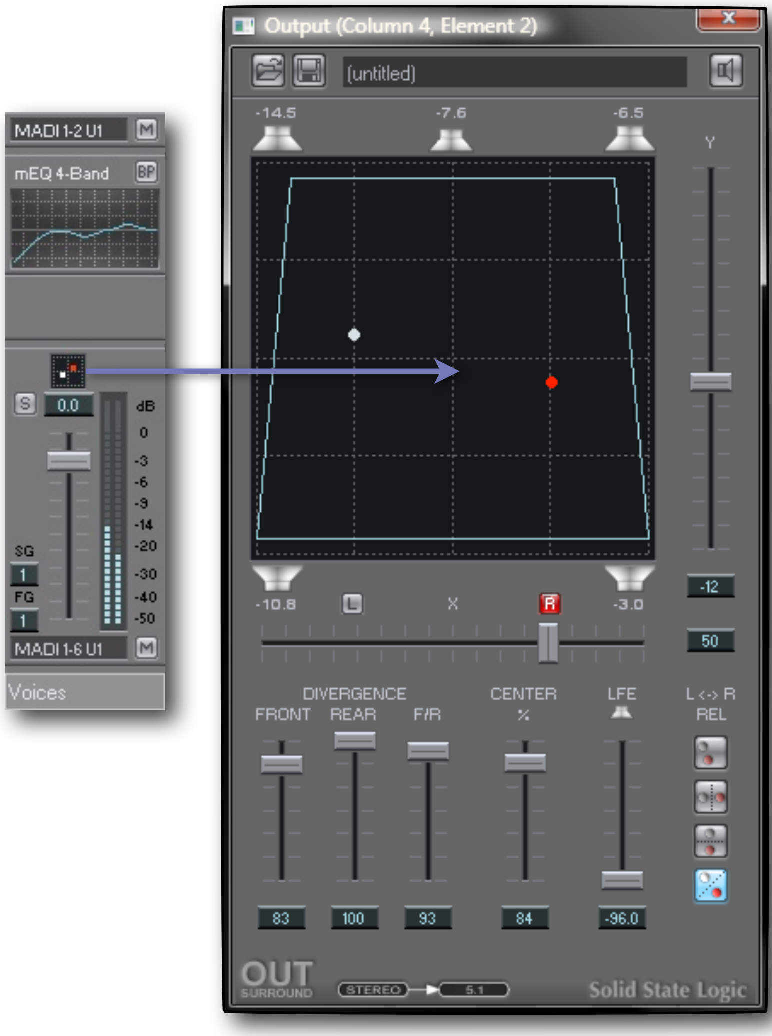
mono to 5.1 panner



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.

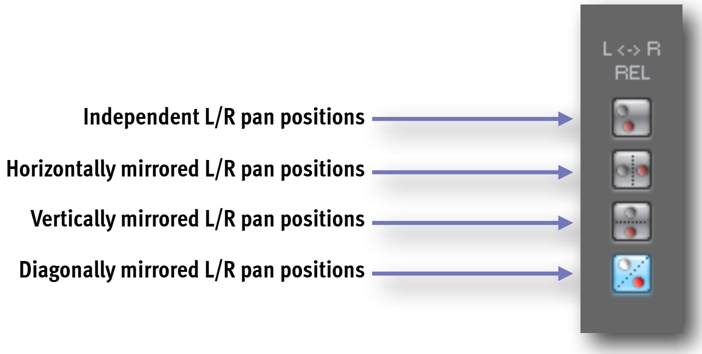
The **LFE (dB)** level control (for 5.1 output panners only) determines the level of the signal routed to the LFE channel (pre output fader). Clicking the "Speaker" icon above the LFE fader will mute/unmute the LFE output channel (the LFE fader appears dimmed when the LFE channel is muted).

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.



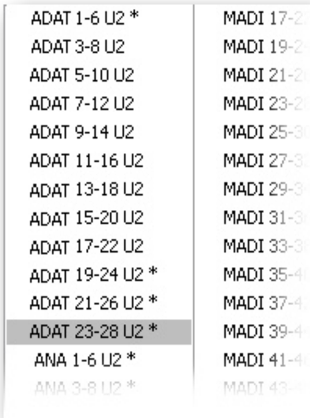
The output routing of the **mono to LRCS and stereo to LRCS** output panners is **Left, Right, Centre, Surround,** so that the left and right signals are on adjacent buses or outputs, in case **just the stereo signal is required.**

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** (e.g., the ADAT5-10 U2 option uses the ADAT A and ADAT B connectors).

Some of the options combine ADAT outputs with analog outputs, or AES/EBU outputs with analog 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

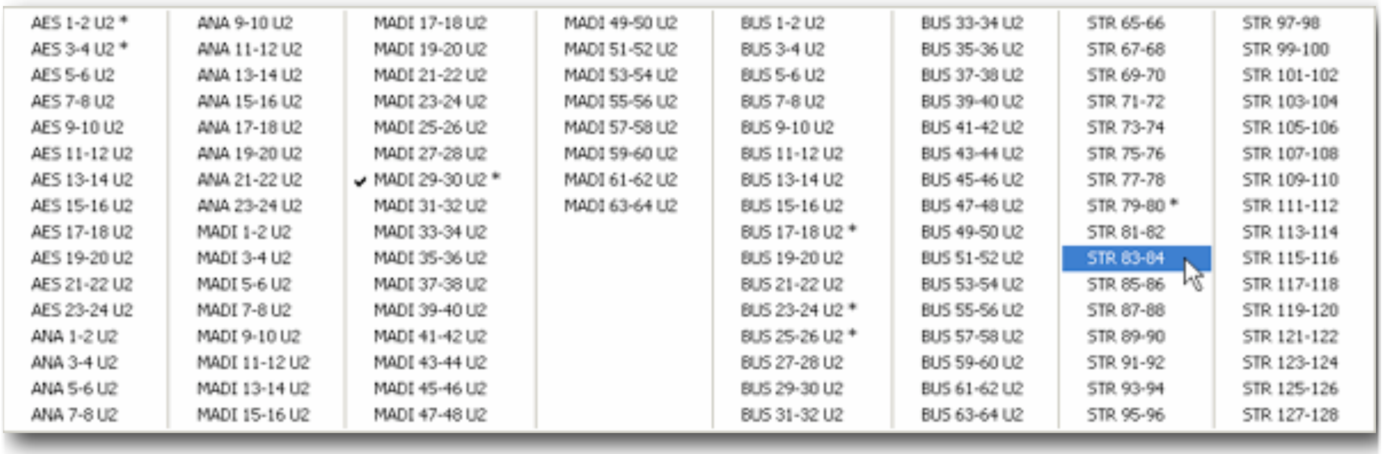
In addition to all that has been mentioned above, mixer columns can also be created with 4, 6 or 8 inputs and outputs. These could be used for example to send and receive multitrack submixes to and from 8 Channel (ADAT/TDIF) ports.

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 **Mixer 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.

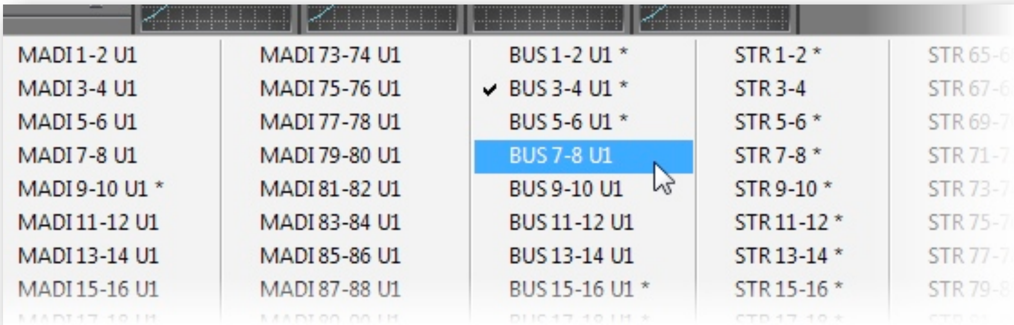


NOTE: The inputs and outputs, buses and streams that are already in use are indicated by an asterisk: (*).

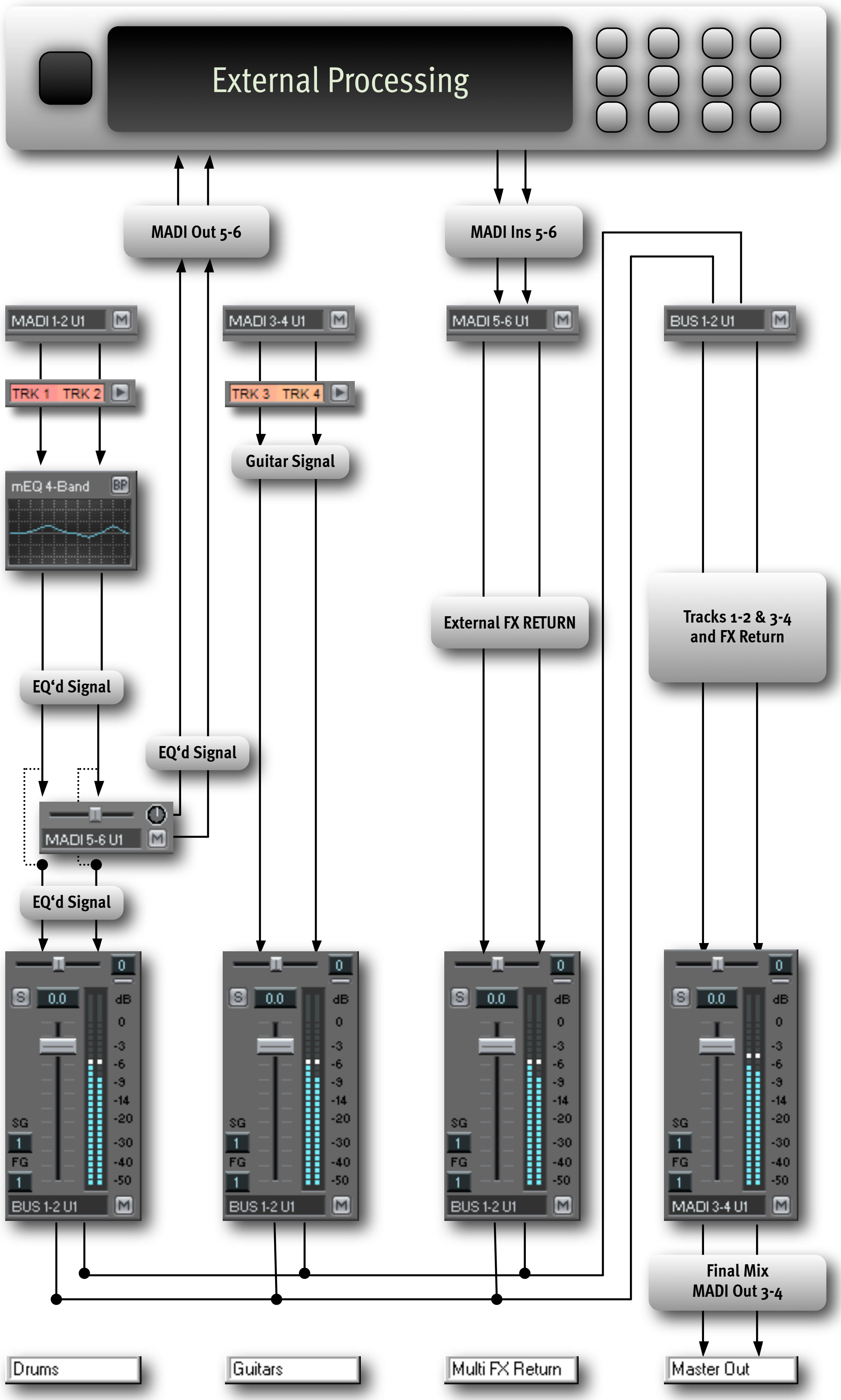
There are no restrictions to how you can use the inputs, outputs, buses or streams, and every signal routed to an output or bus is mixed in equal proportion (after the output or send fader). For instance, you could have 8 channels with MADI 1-2 U1 (or Bus 1-2) as the input, and Bus 3-4 as the output routing. You could then use different EQ settings to split the signal, process it with different effects (e.g., 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.

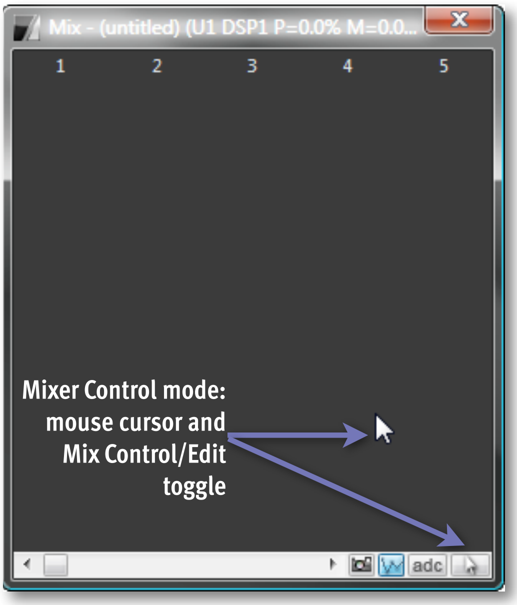


External Processing
MADI Out 5-6
MADI Ins 5-6
MADI 1-2 U1
MADI 3-4 U1
MADI 5-6 U1
BUS 1-2 U1
TRK 1 TRK 2
TRK 3 TRK 4
mEQ 4-Band
Guitar Signal
External FX RETURN
Tracks 1-2 & 3-4
and FX Return
EQ'd Signal
EQ'd Signal
MADI 5-6 U1
EQ'd Signal
BUS 1-2 U1
BUS 1-2 U1
BUS 1-2 U1
MADI 3-4 U1
Final Mix
MADI Out 3-4
Drums
Guitars
Multi FX Return
Master Out

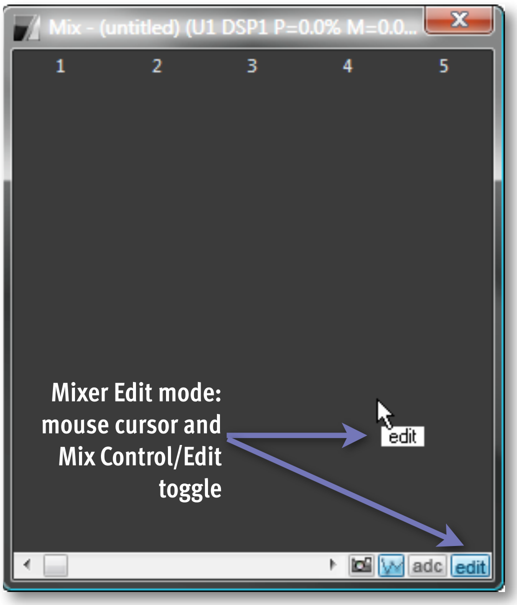
Creating a new Mixer

To fully understand the mixer creation process, it might be useful to think of a particular mixer strip, then set out to create it. You can follow this step by step example to create a stereo mixer strip with a stereo track insert assigned to streams 9-10 and a 4 band multi EQ.

Click **"New Mixer" under the File menu**. If a Mixer is already open, you will be given the option to save it.
A completely blank mixer will appear:

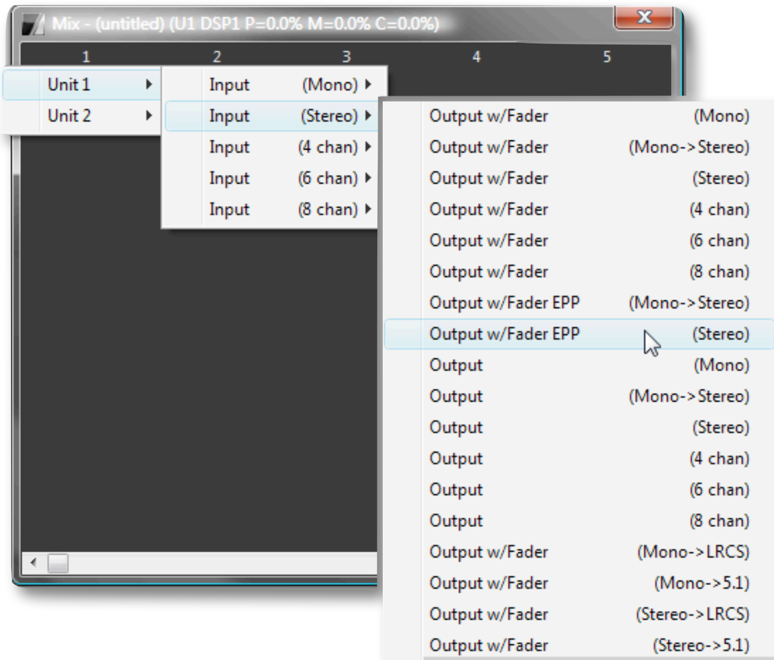


Now press the **Edit toggle button** in Mixer window or press [E], now in Mixer Edit mode the mouse cursor shows "edit" as well as the Edit toggle button:



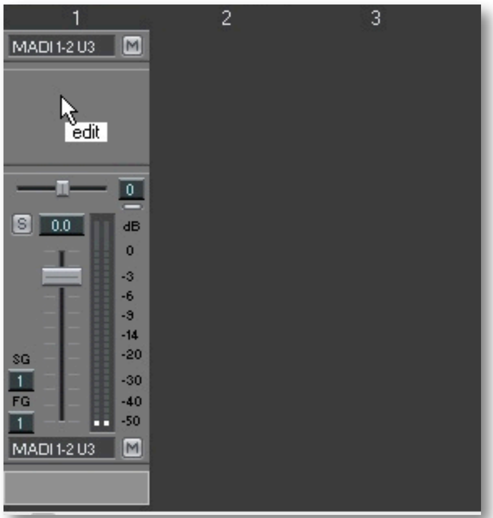
Select the **Create tool** by clicking its icon with the mouse.

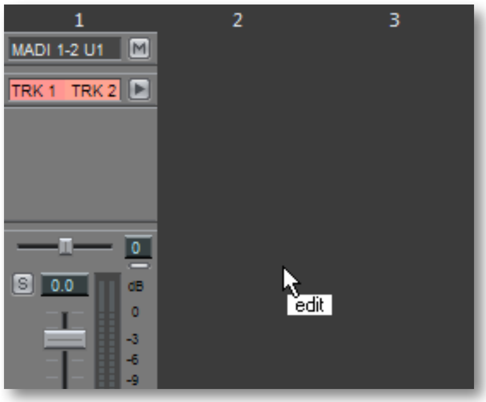
Click **in a mixer column with the Create tool**, hold the mouse button down, select the required type of mixer strip and release the mouse button.
In this example we have opted for Stereo Input/ Stereo Output with Fader and Equal Power Panning in mixer column 1.



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.

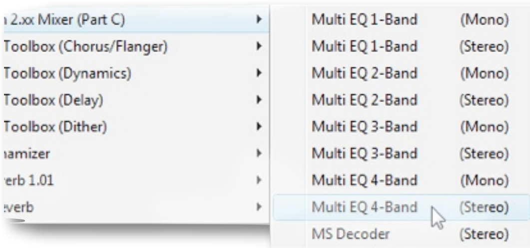
There is also a blank space at the bottom of the mixer strip that can be used to name and colour code the Strip (Scribble Strips).





Still using the Create tool, click within the blank area of the mixer column. A menu will appear where all available mixer elements are listed and can be selected. Select “Track (Stereo)” and release the mouse button to create a stereo track insert.

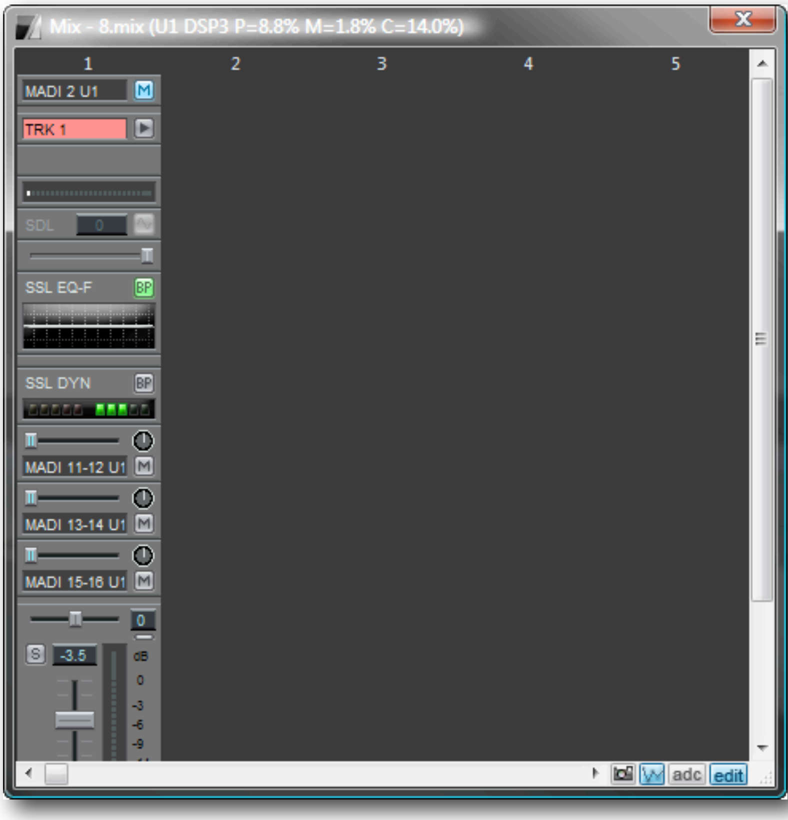
Next, **select the Create tool again,** click in the blank area of the mixer strip under the track insert. The mixer elements menu that you used to create the track insert will appear again.
This time, select "multi EQ 4-Band (Stereo)", then release the mouse button.



The Multi EQ module is now inserted.
The strip is simple but it is now complete.
In Mixer Edit mode, take some time to experiment with the other tools. It's possible to move or copy any mixer element within a mixer strip using the Move or Copy tool.
It's also possible to move or copy an entire mixer strip to another mixer column. If the destination column is already occupied by a mixer strip, that mixer strip and any other mixer strips will automatically be shifted to the right to make room as necessary.
Clicking on the input or output element with the I/O Assign tool will allow you to select a different input source or output destination. With the Delete tool, you can delete any mixer element or strip by clicking on it.

Placing mixer elements into the mixer strip
Clicking with the **Create tool** in an existing mixer strip opens a menu displaying all mixer elements that are available to insert.
The selected element is inserted as soon as the mouse button is released.
If you have followed the step by step mixer strip creation procedure described earlier above, you are already familiar with this menu. There is no limit to the number of elements that can be inserted in a mixer strip (within the available DSP power).

If you insert too many elements and the height of the strip cannot be fully displayed in the SSL Mixer window, a vertical scroll bar will appear.
It is also possible to resize the window. 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.
On the right you see the top part of a mixer strip that contains a mono peak meter, a sample delay line, a 4-Band multi EQ, a SSL dynamics, and 4 send elements.
The window has not been resized while these plug-in elements were inserted, therefore a vertical scroll bar appears on right hand side.



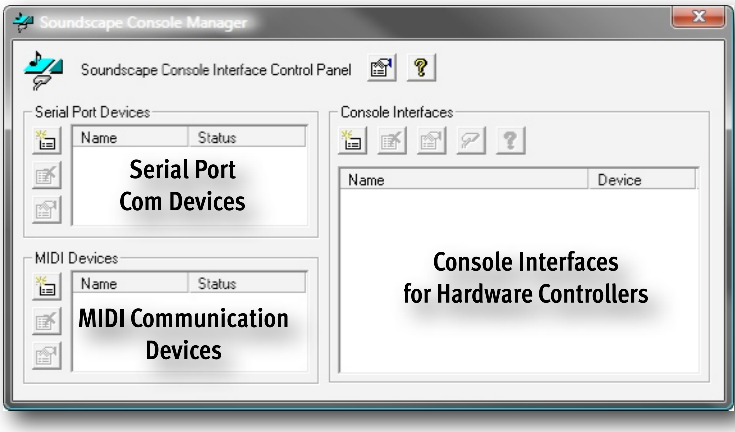
Signal routing in the mixer strip
The signal routing is always from the top to the bottom of a mixer strip. For instance, if you place a track insert below an EQ, this is the position at which audio will be recorded or played back (i.e., if an audio sequencer records from this particular track insert, it will record the equalized signal).

Using the internal buses
The Mixer has 64 internal digital buses for each hardware unit (MX4 or Mixpander) 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.

Using Console Manager and a Hardware Control Surface

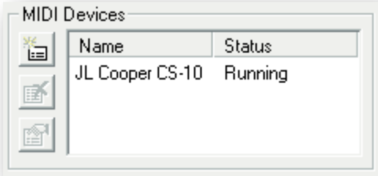
Console Manager Properties Window
If you have installed Console Manager **(see Chapter 2)** Click on Properties or double click the task bar icon to open the main Console Manager Properties Window

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.



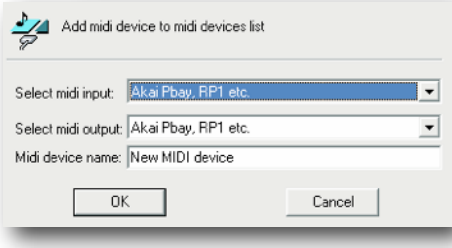
The two sections **on the left** are used to select the Hardware Ports (Serial Com Ports, Midi Interface, USB MIDI) that belong to the Hardware Controller or where the Controller is connected to (MIDI, RS422 etc.).

Setting Up Midi Controller Devices



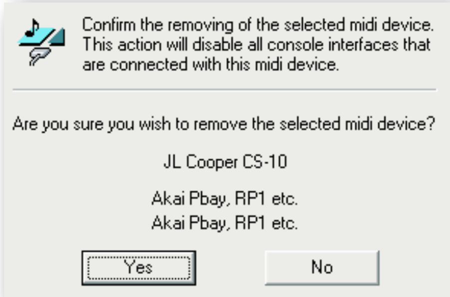
The **MIDI Devices** section has three buttons, **Add, Remove,** and **Properties**.
The latter two appear greyed-out if no Midi device is selected or present.

Add button



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.

Remove button

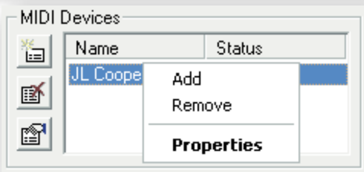


Clicking the Remove button when a device is selected opens a dialogue box where the removal of that device must be confirmed. Click **Yes** to remove the device, click **No** to abort the operation.

Properties button

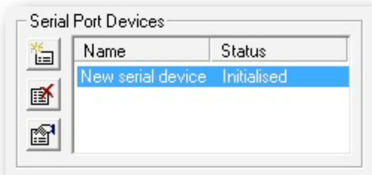
Clicking the Properties button when a device is selected opens a dialogue box which shows the properties of that device, as it was defined upon creation.

Right Click Menu



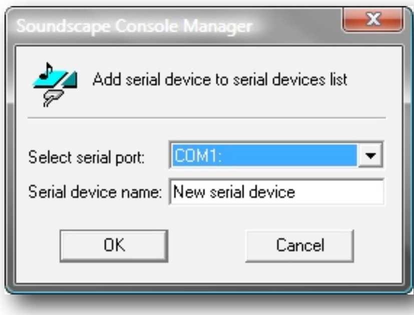
Right Clicking the entry of a MIDI device opens a menu with the same functions as the three buttons described above.

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.

Add button



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.

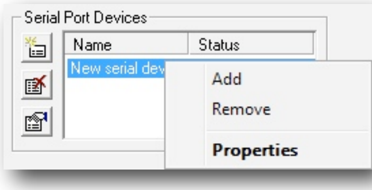
Remove button

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.

Properties button

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.

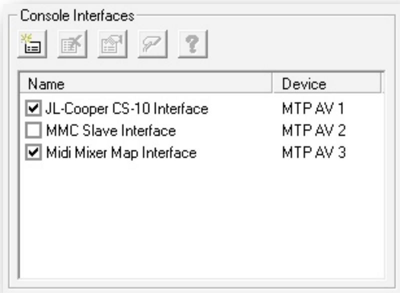
Right-Click Menu



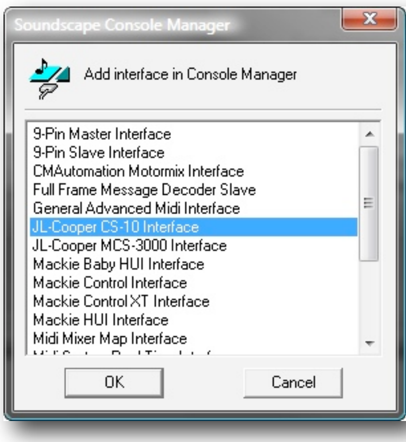
Right Clicking the entry of a Serial Port device opens a menu with the same functions as the three buttons described above.

Setting up Console Interfaces

The **Console Interfaces** section has five buttons, **Add, Remove, Properties. Change Device** and **Information**.
The latter four appear greyed-out if no Console Interface device is selected or present.



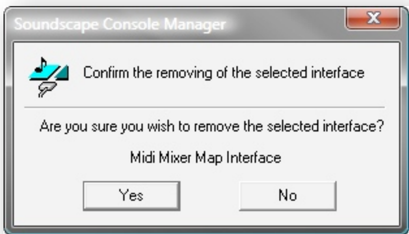
Add button



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.

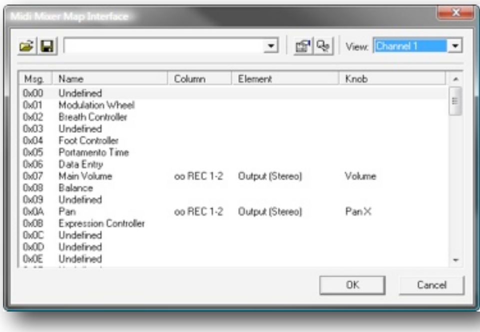
Select the appropriate Protocol/Device Interface and press **OK.**

Remove button



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.

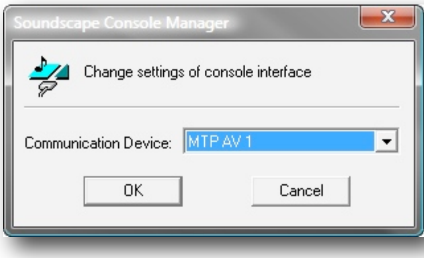
Properties button



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.

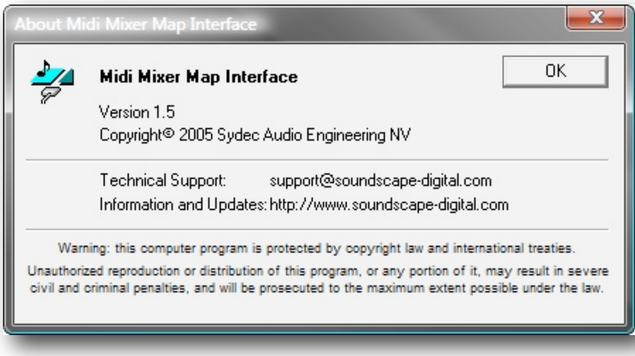
Change Device button



Clicking this 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:

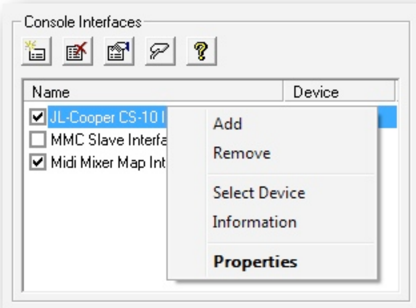
NOTE: While any controller device can be selected from the list, the selected Console Interface will only function correctly with a compatible device.

Information button



Clicking this button opens an information window about the selected Console Interface.

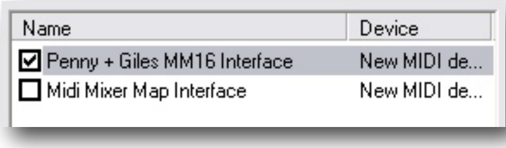
Right Click Menu



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 the Console Manager configuration

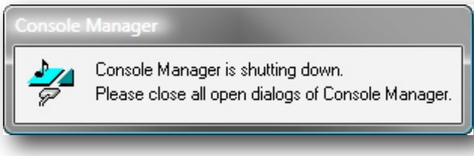
Activating the option **Connect to Console Manager** in the menu: **Settings|Preferences|Mixer** initiates the basic communication betwen the SSL Soundscape V6 and the Console Manager Module.
Performing a **Save Settings** in menu: **Settings|Save Settings** while Console Manager is running and connected, makes the connection permanent and Console Manager will be launched automatically the next time the SSL Mixer is started.

Console Managers 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 V6) 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 SSL Soundscape V6 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.

Standard DSP Mixer Plug-Ins

All mixer elements that can be inserted into a mixer strip are called "plug-ins".
These can be the standard elements and plug-ins included in the SSL Mixer software or optional plug-ins, in many cases developed by third party companies: any element can be inserted anywhere in the Mixer, before or after any other element. This sections 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 Mixer Edit mode to Mixer Control mode, click on the Mix Control/Edit Toggle button in the lower right corner of the mixer window or press [E].

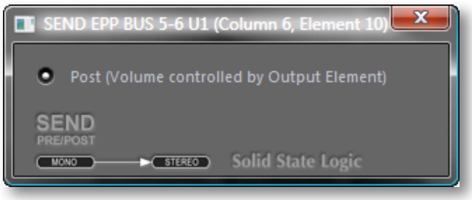
Using the various mixer elements is straightforward. Some general guidelines are provided below which are valid for most mixer elements:

- **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 procedure but over the fader's value box.
- **Pots** (e.g., 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 dialog boxes where the required value can be entered (e.g., 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 dialog 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 (e.g., the "MUTE" buttons in the multi EQ window).
- **"Nodes"** can be clicked and dragged with the mouse, for example to shape an EQ curve in the a multi EQ window, or to place a sound in a surround panner sound field window.
- **Standard Windows option boxes and check boxes** are sometimes provided to select or activate/deactivate certain functions (e.g., to select rectangular or triangular dithering and activate or deactivate noise shaping in the dither element dialog 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 dialog boxes.

Sends

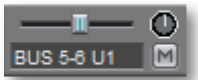
Several types of send elements are available: mono, stereo (with a balance control), mono to stereo (with a pan control), with or without Equal Power Panning.
Any number of sends can be inserted anywhere in a mixer strip as required.
Any internal bus, external output or stream can be selected as the send destination using the I/O Assign tool (in Mixer Edit mode).
All send elements have a mute button.

Double-clicking on a send element opens the following dialog box:

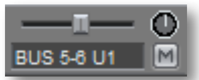


- **Pre-fader sends** (box unchecked) do not respond to changes made to output element settings, regardless of the output element's position in the mixer strip.
- **Post-fader sends** (box checked) respond to changes made to the fader, mute button and solo button settings of all outputs placed below them, except where these outputs are separated from the send element by an input element.

Please note the light blue identifying marks present on Pre-fades sends that allow to see the pre/post status of a send from a glance.



Pre Fader



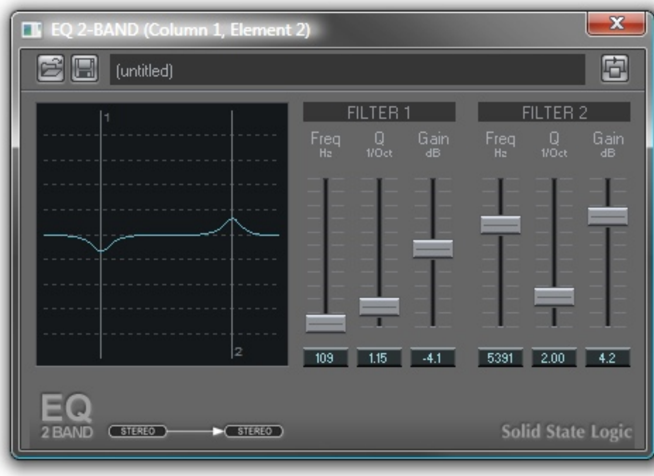
Post Fader

If you want to use a send element as an **auxiliary send to an external effects units,** set the routing of that send element (using the Track Assign tool in Mixer 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 or a feedback loop will be created!), 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

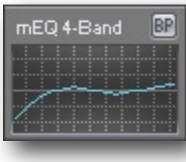
The 2-band EQs are fully parametric, allowing band pass or notch filters with variable Q (bandwidth), cut and boost and centre frequency. You can place as many EQs as you need in a mixer strip to add more bands.
The Quality Factor (Q) determines in a passive filter how selective (narrow) the filter is.
High Q means high quality and therefore 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.



The EQ parameters can be adjusted in real-time by using the mouse buttons in the EQ parameters value boxes in the mixer strip or by using the scrollwheel. 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 dialog box where the value can be entered with the keyboard.

The mono 2-band EQ element has two separate fully parametric sections and for the stereo 2-band EQ, there are in fact four linked active EQs, providing two bands for each channel.
Double click 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, mono or stereo. As usual, more than one multi EQ 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.

Below the graph display, on the left hand side, an **input attenuator** can be used to lower the input gain when boosting certain frequencies in the EQ that would otherwise cause overloads.
If the signal overloads, the EQ curve will change from blue to red (also in the little Mixer Graph).
Use the Input attenuation to decrease the Input volume .
Click and drag the fader, click at the desired fader position, or use the value box to adjust the input gain.

On the the right of the input attenuator, four parameter fields (**"Type", "Frequency", "Q-factor", and "Gain"**) allow flexible manipulation of the currently selected EQ band. On the right you see a 4-band stereo EQ, the band tabs to the EQ band (Band 1 selected). Each band also has an individual "MUTE" button.

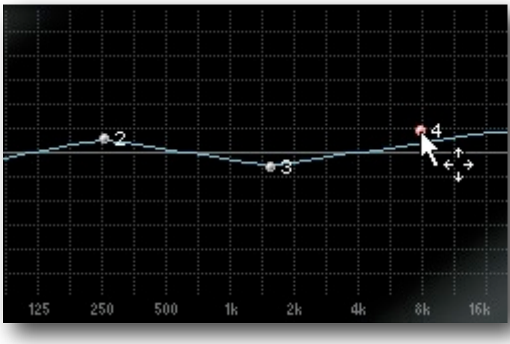
The **"Type" parameter** offers eight different options: low pass, high pass, band pass with constant skirt gain, band pass with constant peak gain, peaking EQ, notch filter, low shelving, and high shelving.
The icons on the buttons show the resulting curve of these EQ types, to make yourself comfortable with the EQ types you can also hover with the mouse over a button until a Tooltip with the Type description appears.

"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 little up and down buttons inside the value boxes.
To enter a parameter value with the keyboard simply double-clicking on a value box.

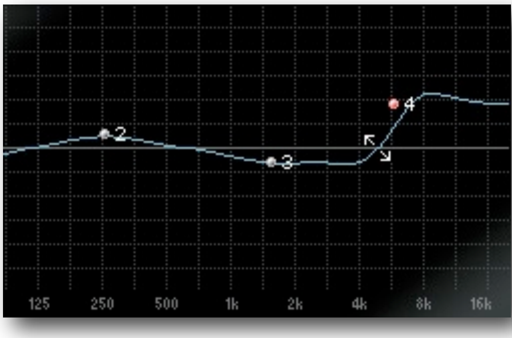
Left-clicking the "MUTE" buttons toggles between mute and unmute of the EQ band. The button is highlighted when the EQ band is muted.
Right-clicking a "MUTE" button "solos" the corresponding EQ band, meaning that the other EQ bands become inactive/muted. When an EQ band is "soloed", right-clicking the "MUTE" button a second time returns all the buttons to their previous state, whereas left-clicking deactivates all EQ bands.
Right-clicking in the EQ band tab near the "MUTE" button 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.
- The curve itself can also be shaped by clicking and dragging, to increase or decrease the bandwidth of an EQ band.

Dragging a node:



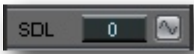
Dragging the line:



The Top bar of the Multi EQ contains the Standard Preset Load/Save and Bypass buttons.

Sample delay line

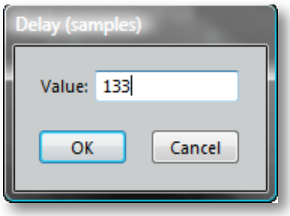
The sample delay line element can be used to compensate for time delays generated in the mixer or resulting from AD/DA conversion. It can also be used to invert the phase of a signal.



Phase inversion button

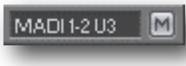
Delay value (number of samples)

The signal can be delayed by up to 255 samples.
The value can be edited directly in the value box using the mouse buttons or scroll wheel in the usual way. Alternatively, double-clicking inside the value box opens a text entry dialogue.
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 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 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.

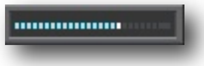
Faders

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.

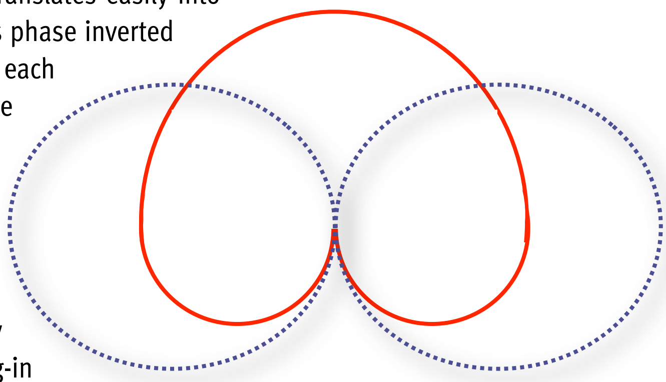
Mono peak meter



Stereo peak meter

M|S Decoder

The MS ("mid-side") recording technique combines two signals in a matrix, typically from one cardioid and one figure of eight condenser microphones, to create a stereo signal. This signal translates easily into mono (the signal recorded from the figure of eight microphone is phase inverted on one side, summing into mono the left and right sides cancel each other, while the signal recorded from the cardioid microphone remains unaffected.
It is also possible to control the stereo spread by changing the relative levels of the mid and side components.



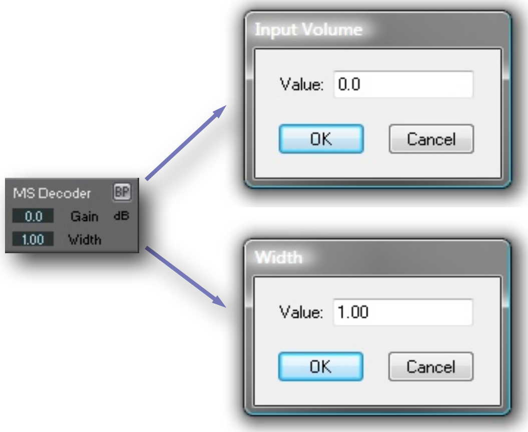
Cardioid microphone, mid signal
Figure of eight microphone, side signal

On a traditional mixing console creating an MS matrix sometimes requires several mixer strips. In the SSL Mixer, only one stereo in/stereo out mixer strip with the MS Decoder plug-in element is needed to create the stereo pair, and dedicated controls allow easy adjustment of all parameters.

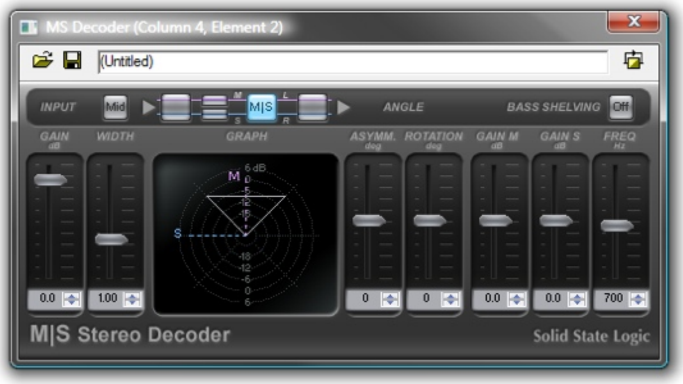
The **gain** and **width** parameters for the MS signal can be adjusted directly in the mixer strip using the mouse buttons or scroll wheel.

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-click on the mixer element to open the main MS Decoder window.



Input section

Lock Mid button

Lock Mid activated

The relative gain of the mid MS matrix input signal is not affected by the "width" setting .

Lock Mid not activated

The relative gains of the mid and side MS matrix input signals vary at the same time and in opposite directions.

Gain (-36.0, 6.0dB)
The overall input gain applied to both input channels.

Width (0.00, 3.00)
Ratio expressing the gain of the side MS matrix input signal divided by the gain of the mid MS matrix input signal.

Matrix/Routing section

Swap Inputs
Swaps the input channels before the signals enters the MS matrix .

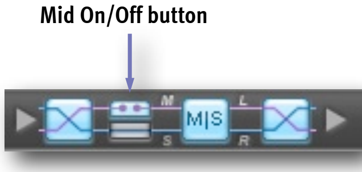
Swap Inputs button

Swap Outputs
Swaps the output channels post MS matrix .

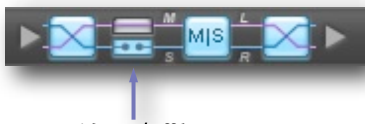
Swap Outputs button

Toggle Mid On/Off button
This button cuts the mid MS matrix input signal, so that the effect of the MS matrix on the side signal can be monitored easily.

Mid On/Off button



Toggle Side On/Off button
Clicking this button cuts the side MS matrix input signal, so that the effect of the MS matrix on the mid signal can be monitored easily.



Side On/Off button

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.

MS Matrix On/Off button

NOTE: If the MS matrix is active, both input signals are attenuated by 6 dB. Then they are subtracted and summed extract the left and right output signals.

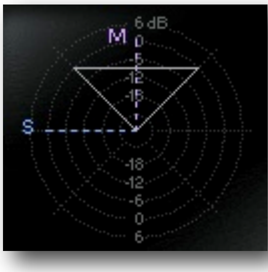
MS Graph

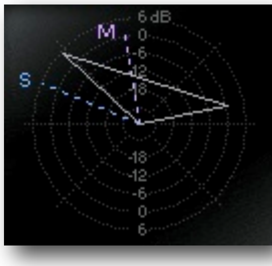
The graph display provides a visual representation of the perceived sound field, as defined and modified in real time by the parameter controls, and contributes to the intuitive operation of the MS Decoder.

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)

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)

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.

The Q-factor of the shelving filter is constant at 0.7.

Gain S (-6.0, +6.0 dB)

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 the shelving filter is constant at 0.7.

Frequency (200, 1300 Hz)

This parameter determines the cut off frequency for both 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

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 combine DSP power efficiency with excellent audio quality. They 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 dialog 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).

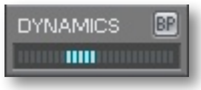
In the main section of the window, the "Mix" fader controls the amount of signal that is delayed and sent to the LFO, the "Delay" fader controls the length of the delay, and the "FdBk" (feedback) fader controls the proportion of the processed signal that is sent back the input.

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.

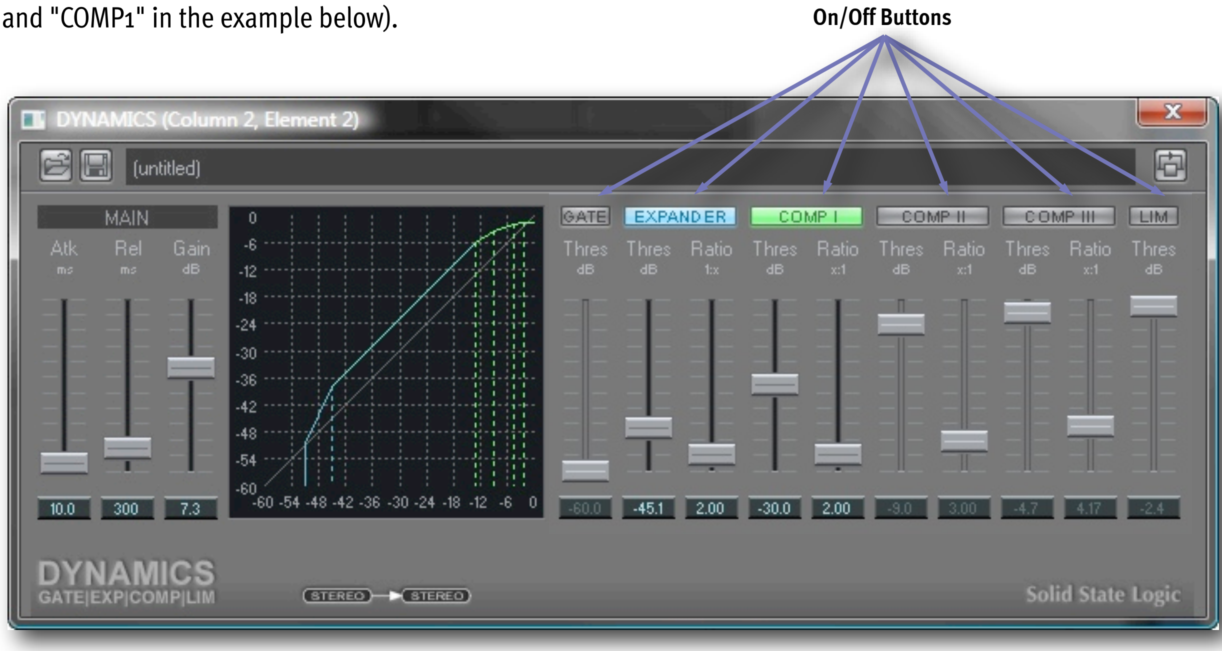
The Chorus/Flanger can produce anything from a slight thickening or doubling of the sound, to the common jangly effect that's great for guitars. Flanging is at the extreme end of the control range, with large pitch changes, delay variations and amazing stereo width and movement. It's great for Sound 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 buttons. Active processing stages appear illuminated.
("EXP" and "COMP1" in the example below).



Dynamics Processing Functions

1. **Noise 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.
2. **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, for auto-ducking, guitar hum etc or it can be used to 'pump the mix' which is commonly heard in dance music.
3. **Three Node Compressor**- The main compressor section has three separate compressor 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.
4. **Limiter** - The limiter allows you to set a threshold for brick wall limiting and with a zero attack time, 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, also useful for optimising broadcast levels during lay-back to tape.

Each of the four dynamics elements can be controlled via a global Main section. This allows the attack time, release time and output gain to be set. Via these three global settings you can decide 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 natural gain reduction caused by compressing a signal.

Setting the Attack, Release and Gain

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 should be 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 can respond quickly to peaks in the signal. 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.

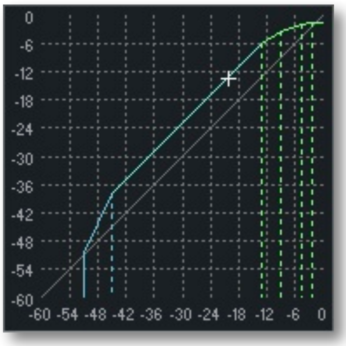
The Attack and Release faders allow you to set the response of the control signal in milliseconds (ms) as required from 0ms to 999.9ms for Attack and 1ms to 9999ms for Release. The Gain fader allows the curve to be positioned in the available dynamic range, changing the output level by up to +/-36dB.

Setting the 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 **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 Comp 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.

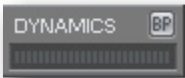
NOTE: The measurement of the input signal depends on the Attack and Release settings and higher values will slow down the animated gain indicator on the curve.

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.

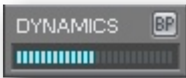


If a threshold fader is moved down and reaches the position of another threshold fader to its left, this fader will also start moving down. However, as long as the mouse button has not been released, returning the fader towards its original position will allow the other threshold settings to be restored.
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



No Processing



Expansion/Gate

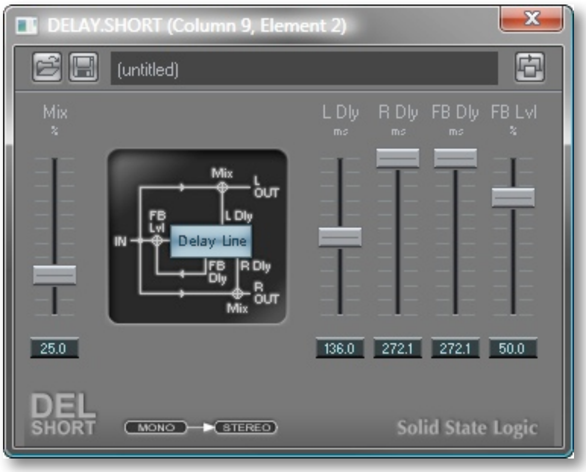
Compression

The amount of gain reduction is shown by the meter in the Dynamics mixer element. The meter calibration is 3dB per segment.
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

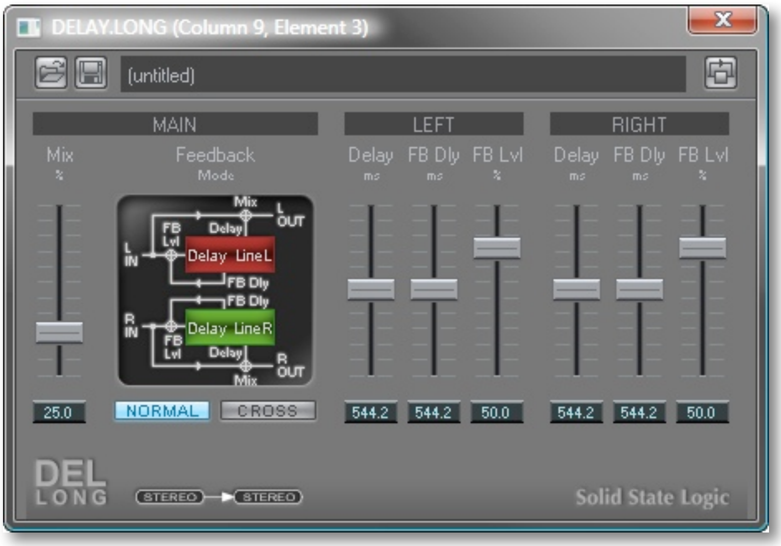
The Delay plug-in provides a 2-tap mono, 2-tap mono to stereo, or 2-tap stereo delay, with selectable cross-linking between left and right channels. Each is available as a Long, Medium or Short delay version, longer possible delays need more DSP Memory, therefore it is advisable to select the element type with the shortest delay time that suits your requirements:

e.g. at 44.1kHz:
Long: up to 1088.4ms
Medium: up to 544.2ms
Short: up to 272.1ms

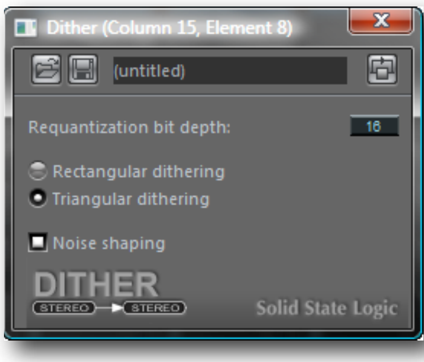


Clicking the "cross" or "normal" button for a stereo delay allow to 'cross couple' the feedback portion of left and right delay outputs for panning delay effects. In cross mode the feedback signal from the left delay line is fed into the input of the right delay line and vice versa.

The stereo elements feature completely independent controls for the Left and Right Delay Lines.



Dither



The Dither element is for use when a lower bit resolution is required than the original audio tracks or following mixing. 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.

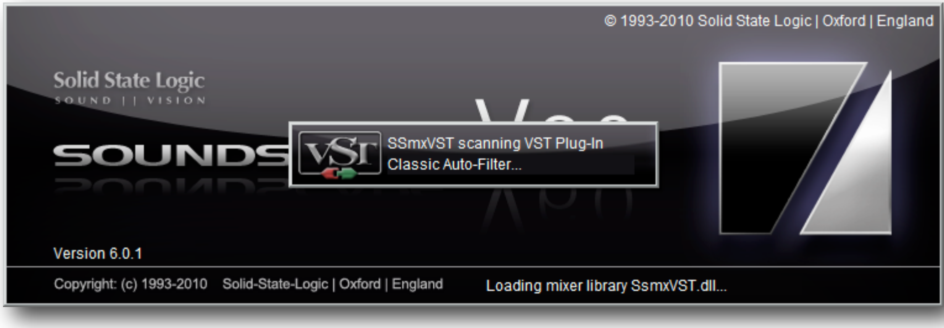
The "Requantization bit depth" can be changed in the value box using the mouse buttons in the usual way. Alternatively, double-clicking in the value box will open a a text entry dialogue.

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 elements.

The VST plug-ins are stored on the PC as files 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 splashscreen where the plug-ins' names are displayed as they are loaded.



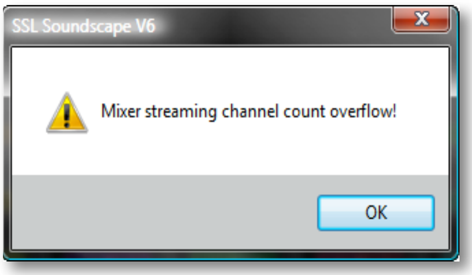
Dialogue boxes may also appear during that phase, for instance when a plug-in requires authorisation.

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.

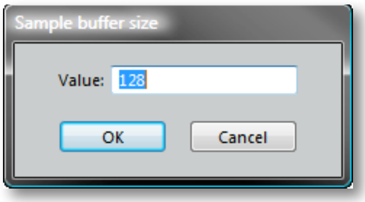
Since the VST plug-ins run on the CPU of the host PC, the audio must be routed at high speed between the SSL Mixer and the PC in order to use VST effects at low latencies.
This is done using the "streams" provided by the MX4 or Mixpander.
One stream is used per channel of audio, so that a mono VST plug-in 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.



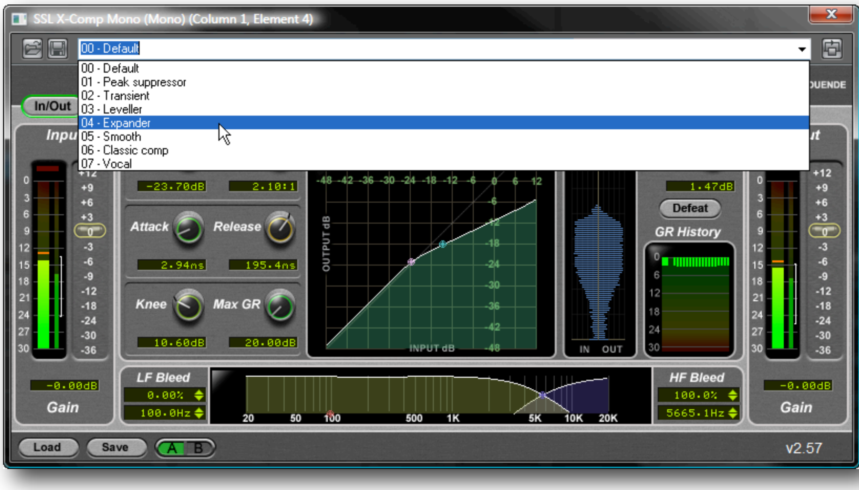
In this case the Mixer will also become inactive until the situation is resolved by deleting one or more VST plug-in(s), reassigning or muting one or more input, output or send element(s) or closing or re-routing a Windows application in order to release some streams.



The default buffer size for native mixer elements is 128 samples. This setting determines the "latency" of the VST calculation.
It can be changed under menu:
Settings|Preferences|Native mixer elements Sample Buffer Size.

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 within the SSL Mixer generally behave in a similar fashion to SSL format DSP-powered plug-ins and SSL Mixer elements in general.



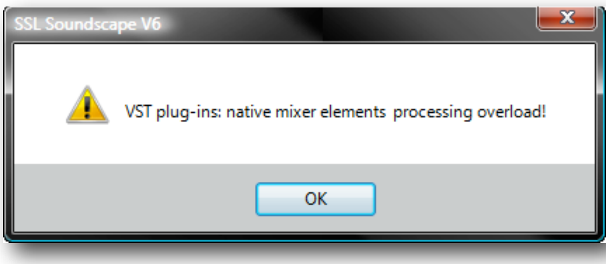
They can be moved or copied using the relevant tools, their settings are saved as part of the Mixer, and automation works in the same way as for other mixer elements.
However, the text box at the top of the main plug-in window behaves a bit differently.
The arrow to its right can be used to open a program 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 a grey background and black text.

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 in the previous section 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.

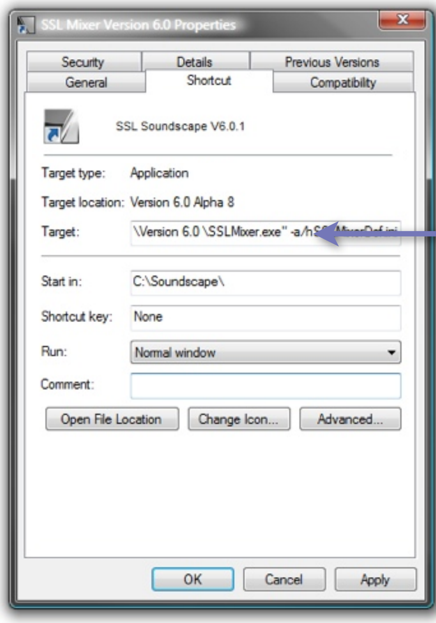
Additional information regarding VST/VSTi plug-in compatibility

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 V6 shortcut's properties.
In a standard installation the SSL Soundscape V6.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"



"-a" command line switch

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 processing of an audio signal takes 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 cumulated processing 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 as necessary to get the output audio signals perfectly aligned. However, this introduces another problem: since all the signals must be aligned with the "slowest" one (i.e., 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.
Therefore, a performer cannot monitor his/her performance in real-time through that system The overall delay is perceived as the inherent "latency" of the 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" (i.e., they present the same overall amount of delay). However, all buses and outputs are not necessarily lined up relative to each other.

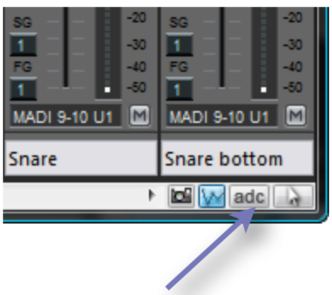
For instance, the following output pairs...:
- **MADI1-2 U1,** used by the "Output (Stereo)" element in mixer column 1
- **MADI3-4 U1,** used by the "Output (Stereo)" element in mixer column 2

...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.

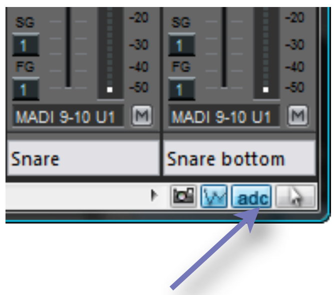
Automatic Delay Compensation toggle [Ctrl]+[Alt]+[D]

The Automatic Delay Compensation can be activated or deactivated by clicking the ADC toggle, which is located in the bottom, right corner of the Mixer window.
Automatic Delay Compensation is active when the button is highlighted.

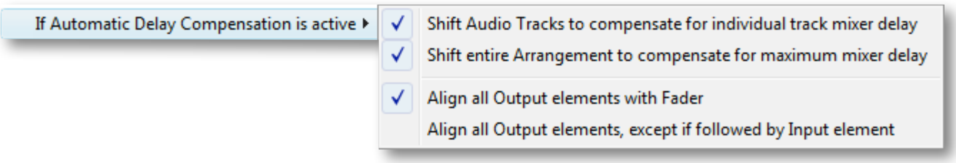
Alternatively, you can click the "Automatic Delay Compensation toggle" item in the **Edit men**u or use the **[Ctrl]+[Alt]+[D]** key command.



Automatic Delay Compensation toggle (ADC disabled)



Automatic Delay Compensation toggle (ADC enabled)



Soundscape V6 offers several options and strategies on how to compensate for processing delays.

The Options are centralised in the menu: Settings|Preferences|Mixer|If Automatic Delay Compensation is active.

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" (i.e. 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

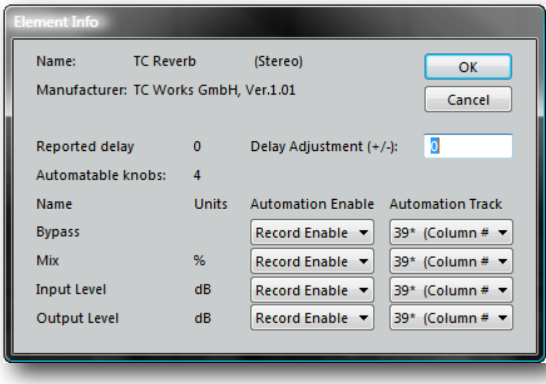
In order to line up outputs that have isolated audio paths in the Mixer, this option can be used.
When selected, all output elements that have an Output fader are lined up. Outputs elements without a fader can be used for signals that should not be lined up (e.g., a send or Column without fader feeding 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.

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 Mixer Edit mode, clicking a mixer element with the Info tool...: ...will open the corresponding "Element Info" window, as shown below 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 the 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 V6 must be restarted for the command line to take effect (settings specified in the .INI file are loaded on startup).

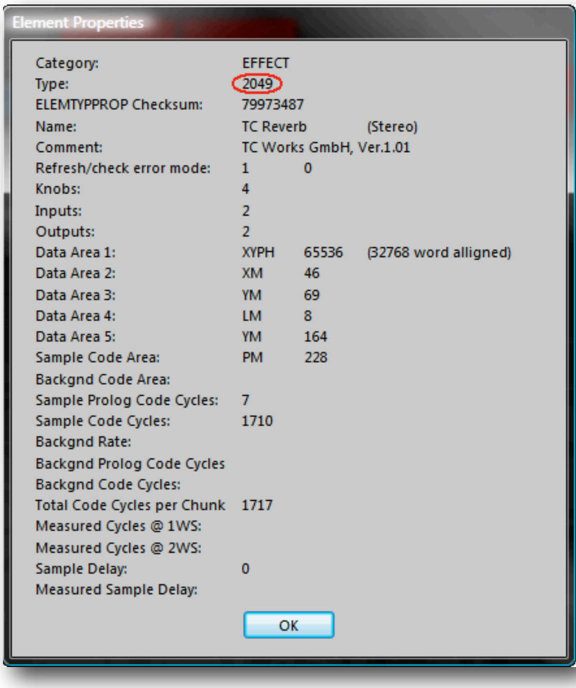
The command line reads as follows:

ElemType XXXXXXXX Delay=Y // Element's name

where:
- "XXXXXXXX" is the unique Element Type ID for the element.
- "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 Mixer 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 V6: 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 **Chapter 3 › Recosring Automation**. 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, but 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.

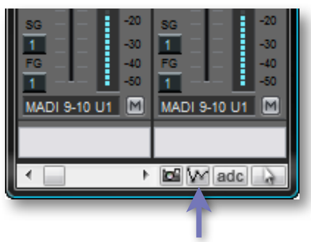
The "Automation concepts and mixer controls behaviour" section expands on certain basic concepts and features of the automation system and their practical implications.

Recorded automation data can be viewed and edited "offline" with three specific tools which will be described in the "Automation editing" section.

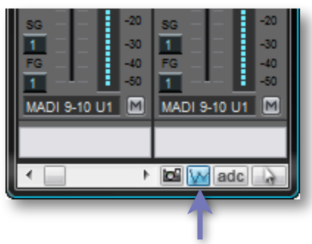
Some vocabulary specific to Automation will be used in this chapter. Automation data is made of "automation events", and these come in two varieties: "continuous events" and "stepped events". The difference will be explained in the "Automation Event Editing tool" section. "Automation events" are generated 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



enabled

Alternatively, you can press [G] on the computer keyboard to enable/disable Automation, but only when the Mixer window is open, or click the "Automation Enable toggle" item under the Edit menu.

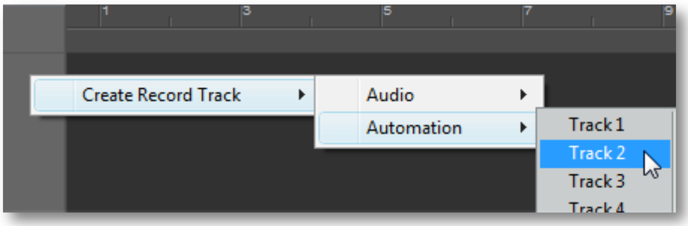
Automation recording

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.

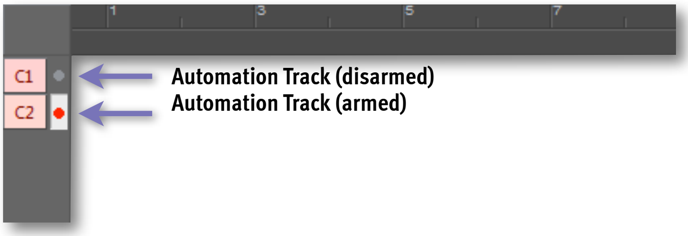
Dynamic automation recording

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.

In order to record automation, an automation record track needs to be created by right-clicking in the Record Track Column:



Then it must be armed using the track arming button. These procedures have been described in detail in **Chapter 3 › Recording Automation.**

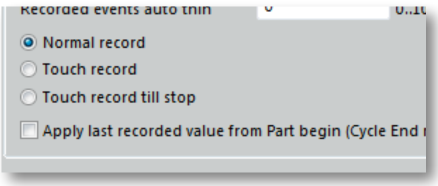


The procedures for audio recording apply to automation recording as well, with a few exceptions. For example, automation tracks cannot be armed from inside the Mixer, or using the computer keyboard. However, loop recording, record loop stacks, manual punch in/out, and the Auto Punch In/Out mode can all 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 (i.e., 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 settings specific to automation recording: "Normal record", "Touch Record", and "Touch Record Till Stop", selectable in the Automation Setup window, which is accessed by clicking "Automation Setup" under the Settings 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

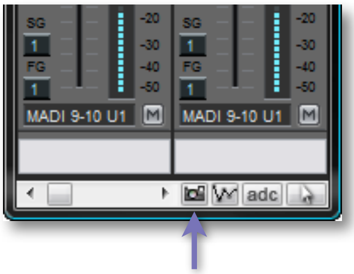
In this mode, the automation events generated during the current pass are recorded, and all automation events from existing active Parts assigned to the same automation track are re-recorded to the new Part as well. Therefore, it is possible to overwrite an existing Part (or section of a Part) on the 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 the Touch Record mode, 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 what happens in the simple Touch Record mode, 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, which is near the Automation Enable toggle in the bottom right corner of the Arrange window, by pressing the [F] key, or by clicking “Snapshot Automation Record” under the Edit menu:



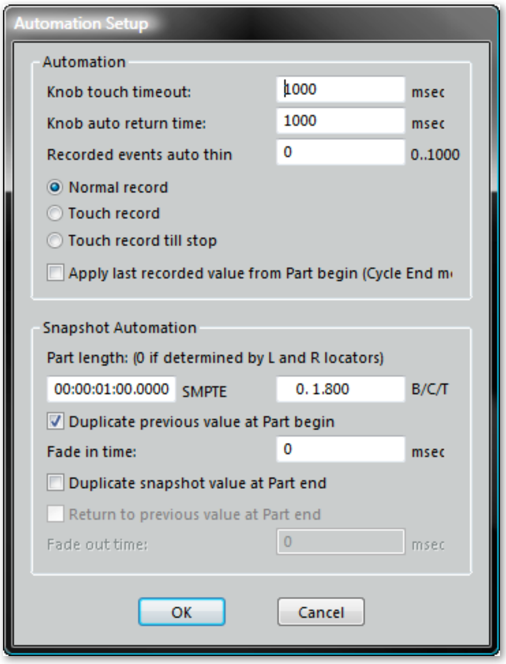
The resulting snapshot shows as an automation Part or group of Parts which starts at the time position where the Snapshot Automation Record button was clicked. One Part is generated for each armed automation record track, which contains information for the setting of each enabled automatable knob assigned 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 mode or Play mode, when a control surface which is not touch-sensitive is used. In this case Soundscape V6 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 as “Knob touch timeout”.
- The “Knob auto return time” parameter is applicable in Touch Record mode 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 controls 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 the "Touch Record" or "Touch Record Till Stop" modes). 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 the 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 controls 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.

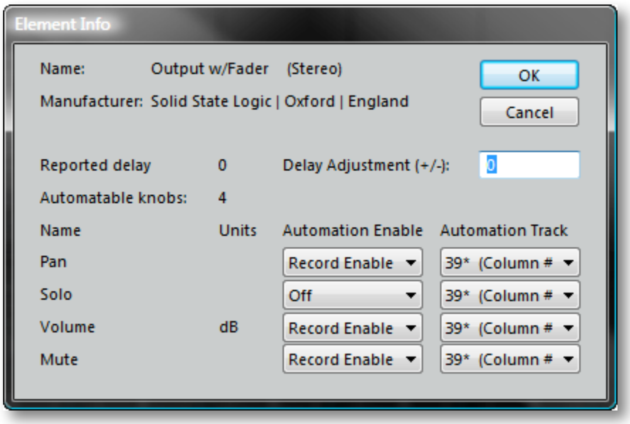
"Gliding" mixer controls and intermediate values

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 mode, 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 (i.e., 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, the “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” mode, 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 are “automate able knobs”, 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” dialog 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.



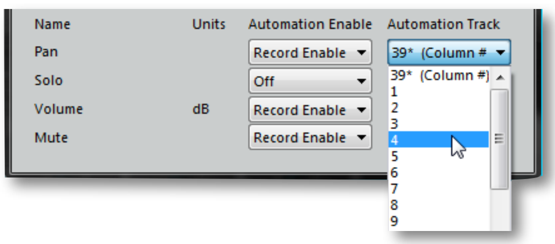
The Info tool allows you to check which knobs an be automated, to enable or disable automation recording and playback for each knob individually, and to assign each automatable knob to any automation Track.
In Mixer Edit mode, clicking on a mixer element with the Info tool will call up the “Element Info” 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. Selecting, for example, “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.

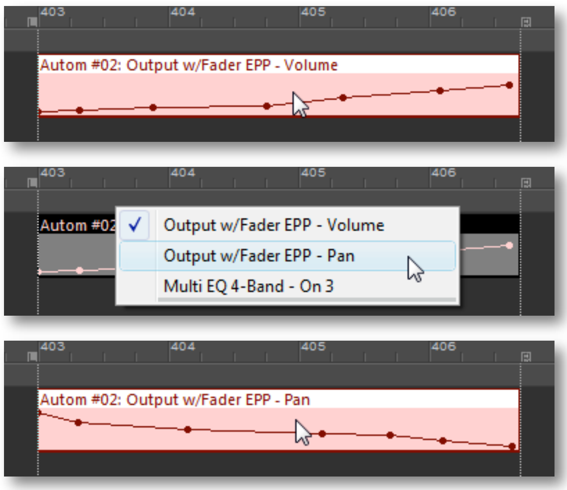
In most cases the default settings are perfectly adequate. You may want to change them for example to save space on-screen by recording automation for several mixer columns on one single track, if you need a lot of virtual tracks for audio Parts.

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 the “Automation Events”, within the Parts, can also be viewed and edited with three specific tools: the “Automation Curve Select”, “Automation Events Thinning” and “Automation Event Editing” tools. They have been mentioned briefly in the “Editing Tools” chapter, but will be examined in detail here:

Automation Curve Select tool [Ctrl]+[Shift]+[A]



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, such an automation Part can, at any time, only display one single layer of automation data, relating to one single knob. 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, 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 [Ctrl]+[Shift]+[E]

The Automation Event Editing tool 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”.

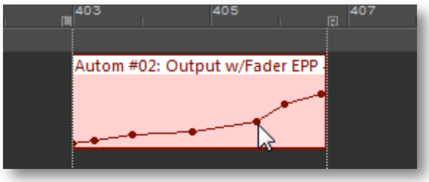
Creating an automation event
Double-clicking in an automation Part with the Automation Event Editing tool will create a new event within the currently displayed curve. The new event’s value will be determined by the vertical position of the mouse pointer, and its time position will be determined by the 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.

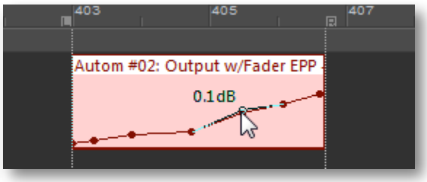
NOTE: If needed, use the Automation Curve Select tool first to display the required curve. Event creation is only possible if at least one event of the same type has already been recorded in the Part (i.e., if there is already an automation curve).

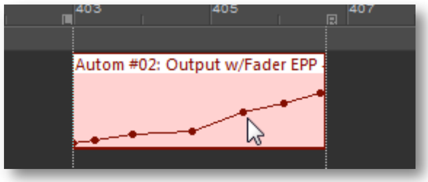
Deleting an automation event
Double-clicking on an existing event node will delete it. This needs some precision and zooming in to an appropriate level will sometimes be helpful (when a lot of events are displayed in a small space on the screen).

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.

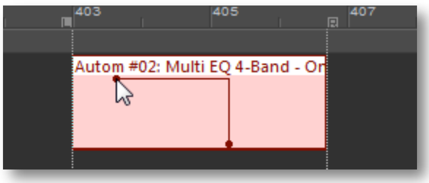
For automation “continuous events” (the kind of events created by e.g. a volume fader or a pan or EQ gain pot, which generate a continuous stream of events with many possible values), the target value will be displayed and updated in real time as you drag the event node, as in the second screenshot below, until you release the mouse button:

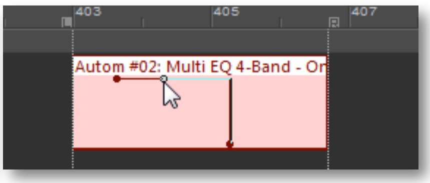


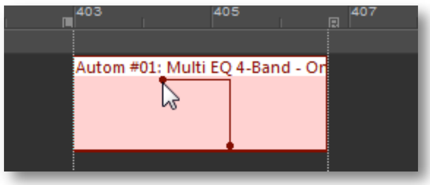




For automation “stepped events” (the kind of events created by a solo, mute, bypass, or phase inversion button, or a group of EQ curve selection switches, all of which can be in one of a limited number of positions), no value is displayed as you drag the event node, and you can only move it to “allowed” 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”).







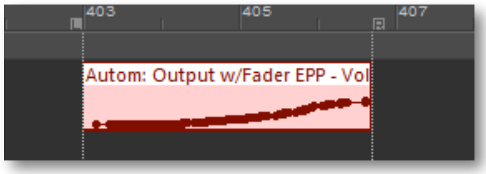
NOTE: The edits carried out with the Automation Event Editing tool result in a new Take being created. In that sense the Automation Event Editing tool acts as a processing tool: each time an event is edited, the corresponding Take is processed. Therefore, this tool is subject to the setting chosen under menu: 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** [Ctrl]+[Shift]+[F]

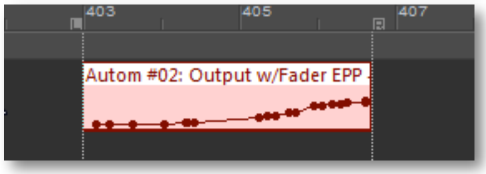
Automation "stepped events" as defined in the previous section do not require thinning, because they are generated one by one exactly as required. Therefore, by design, the Automation Events Thinning tool has no effect on them.

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".

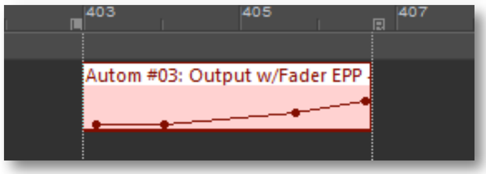
This example shows the recorded automation curve for a volume fader movement performed with the mouse:



This shows the same volume fader automation curve, after processing with a thin factor of 200:



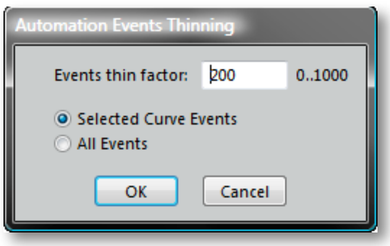
Here, the maximum thin factor of 1000 has been applied to the original automation curve:



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 menu: 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 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.

Click on an automation Part (or group of selected Parts) with the Automation Events Thinning tool. The following dialog box will appear:
Enter he 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" inside the Automation Setup automatically applies a thinning factor to any new automation recording.

Mixer editing with existing Automation

The extreme flexibility of the Soundscape V6 software means that some situations can be engineered that would be considered extreme in a hardware environment; in particular, within Soundscape you can edit the Mixer itself even when some automation data has already been recorded. In the hardware world this would be the equivalent of, for example, adding a mixer column in the middle of a mixing console, including all the necessary rewiring and soldering, or adding a fader in the middle of an existing mixer column - all of this while a session is in progress! This is not only impossible in practice in the hardware world, it is also beyond the capabilities of most software packages, which provide a fixed, “take it or leave it” mixer, that may be “virtual” but is often just as inflexible as the hardware equivalent.

While flexibility is highly desirable, it also means that it is advisable to understand as much about the system as possible. After all, if a technician dropped into your session to carry out an on the spot modification to your hardware mixer, you would hope they knew a thing or two about electronics!

In this light, there are some things you should know if you are going to edit your Mixer when some automation Parts have already been recorded:

When automation data for several mixer elements of the same type (e.g., 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 below:

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 the 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 (it is important not to overwrite the original Part, so please choose an empty virtual track).
- 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.

File Manager

Soundscape V6's File Manager Window is the central place to access any kind of relevant file available to your PC. In a way very similar to Windows Explorer, its purpose is to handle Arrangements, Mixes, Audio Takes and Automation Takes used by the Soundscape Software.

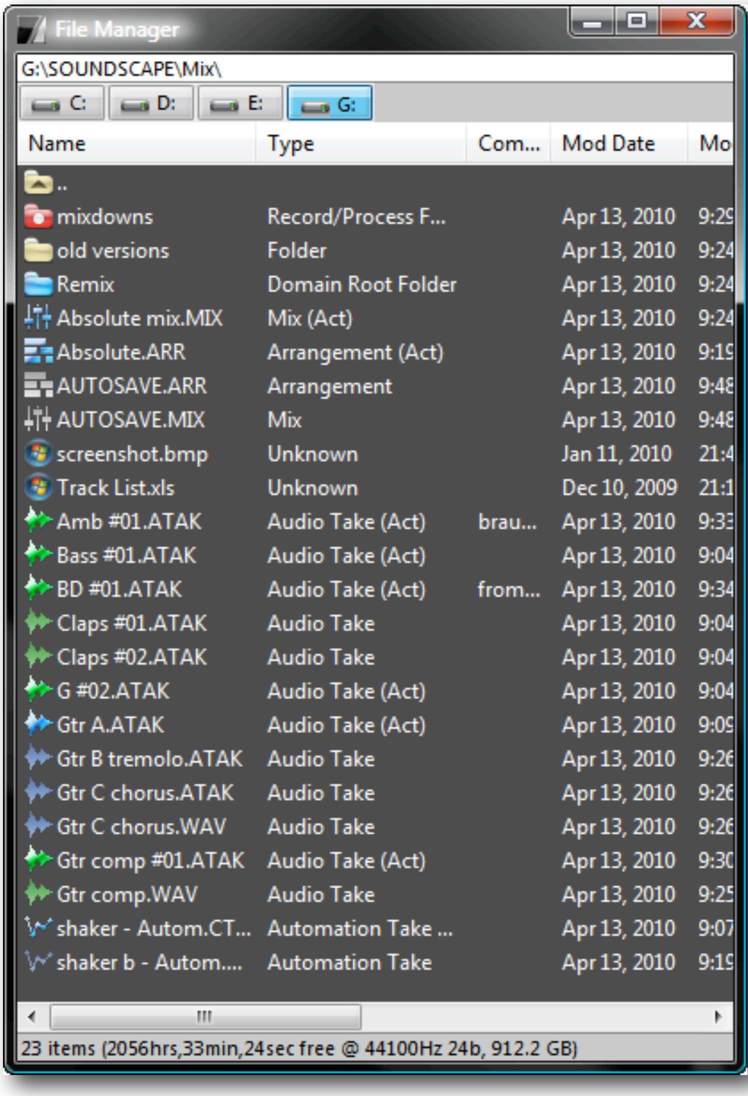
Soundscape V6 doesn't have any concept of a Bin, Media Pool or the necessity to Import directly readable audio files into a collection. The File Manager relies on simple Drag' and Drop action to add audio or automation to the Arrangement.
There is also no concept of Regions or anything similarly confusing 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, collect all project files and create 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.

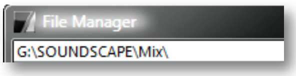
Icons of Takes that are referenced by the Parts in the Arrangement are highlighted in green (24-bit audio) or in blue (16-bit audio and automation Takes).



Audio files and even complete Arrangements can be auditioned directly in File Manager.

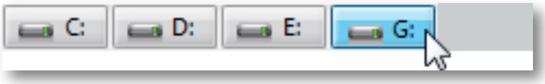
Current Path

The current Path is displayed in the Top of the File Manager Window. 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) .



Drive Bar

In the Drive Bar all logical Drives attached to your system are available.
The opened drive button is illuminated.



The Drive Bar remembers the path of the last opened folder for each drive, so navigating from drive G: to E: and back, File Manager will show the previously opened folder in drive G.

The Drive Bar Buttons show different Icons for internal HDD's, external HDD's (USB/eSata/Firewire), Removeable Media(CD/DVD/Tape) and mapped Network Drives.

File Manager only Lists Drives that have a Drive Letter assigned in Windows Disk Manager.

Soundscape V6 only scans for drives only startup.

Hence File Manager does not automatically update the Drive Bar, when an external Drive is plugged or unplugged.
To update the Drive Bar SOUNDCAPE V6 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 Button.

Hiding Drives

To hide certain logical Windows drives from the Drive Bar (ie. DVD Burner, System Drive) you can use the File Manager Preferences in menu: **Settings|Preferences|File Manager** to activate a Hide Status for any logical Drive letter (from A: to Z:).

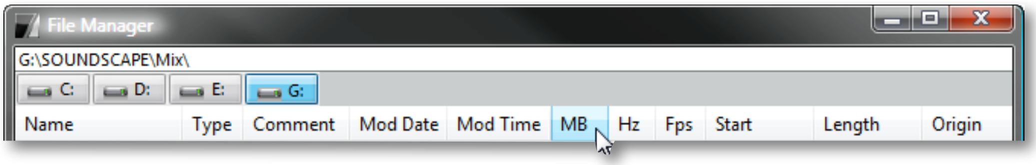
Hiding and Showing drives is immediately reflected in the Drive Bar. Active Takes from hidden Drives remain playable by referencing Parts in the Arrangement.

This Feature is also explained in **Chapter 5 › Settings Menu › Preferences**

Attributes Bar

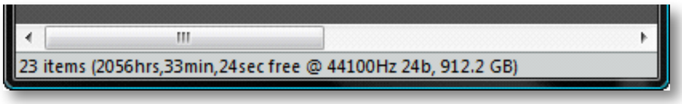
The Attributes Bar shows the attributes column. Clicking on a file attribute allows defining the sorting priority. Clicking for a second time on an item will toggle between ascending and descending order.

Each column can be resized by dragging the left or right extends of the attribute, in spreadsheet style.



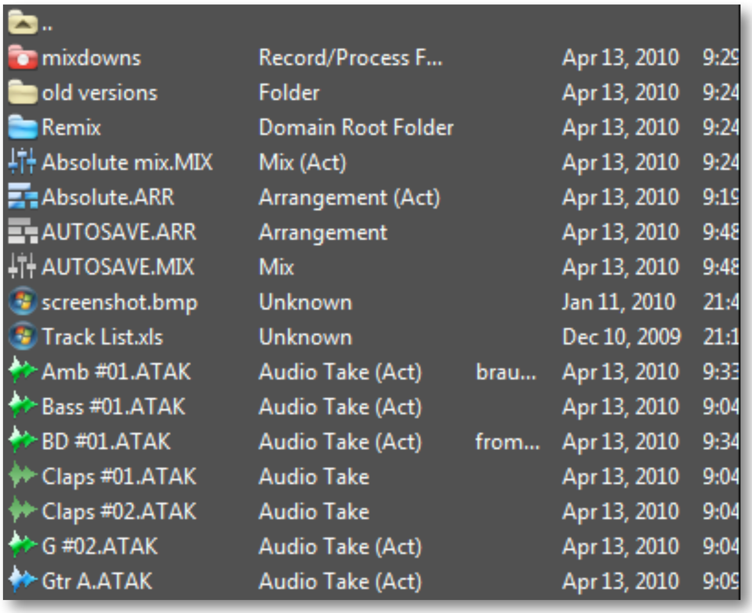
Status Bar

The Status Bar shows the number of files present in the folder or number of selected files and in brackets the remaining free space in GB and in time per mono track at the current sampling rate on the selected drive.



File List View

The File List View shows all relevant information for files and folders and allows all sorts of administration tasks like renaming, moving, copying, pasting, deleting, etc.



File and Folder Operations

The File Manager Window uses context sensitive interaction, using the left and right Mouse buttons.

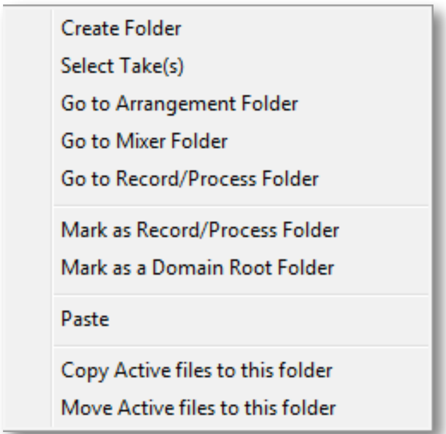
Left Double Click

- On any blank space of the File List goes up one level in the folder tree
- Double Click on this Icon (or this in case the folder is part of a Domain/Root) the File List goes up one level in the folder tree
- Double Click on a Folder opens it

Right Click context sensitive Menus

Right clicking on a File or Folder or a Blank Space inside the File List View opens different context sensitive Menus.

Right Click on an empty Space



Create Folder

This command works in a very similar way to Windows Explorer.
When a folder is created it is ready to be named.

Select Take(s)

If Parts are selected in the Arrange window, this 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.

Go to Arrangement Folder

Opens the folder where the currently active Arrangement is located.

Go to Mixer Folder

Opens the folder where the currently active Mixer is located.

Go to Record/Process Folder

Opens the selected Record/Process Folder for the active Arrangement.

Mark as Record/Process Folder

This 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.

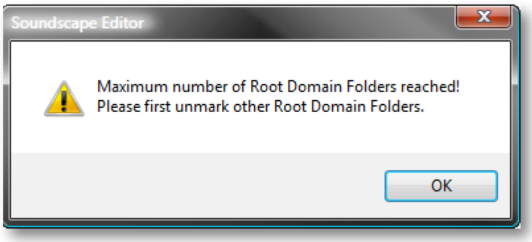
This folder setting is saved in the Arrangement file so every time it’s opened, the location to store audio, automation, mix and arrangement files is remembered.

Mark as a Domain Root Folder

Selecting this option marks the selected or currently open folder as ¨Domain Root Folder¨.
The Domain Root folders allow Soundscape V6 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.

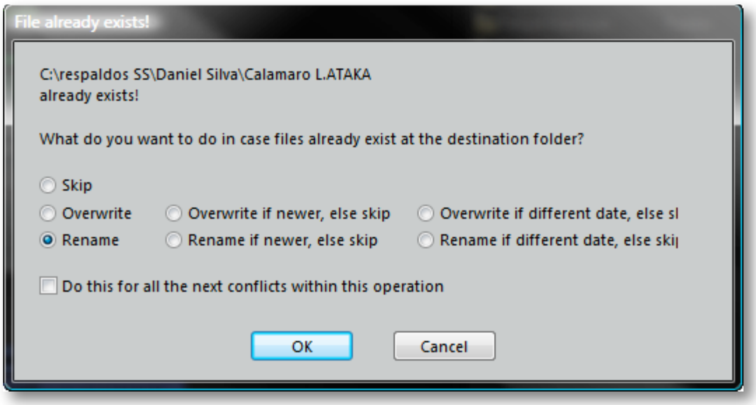
Once you attempt to assign the 9th Domain Root, an Error Dialogue will pop up:



Please note that marking a subfolder of a Domain Root Folder, removes the status from the parent folder.

Paste

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 exisiting 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 File Name Conflict.

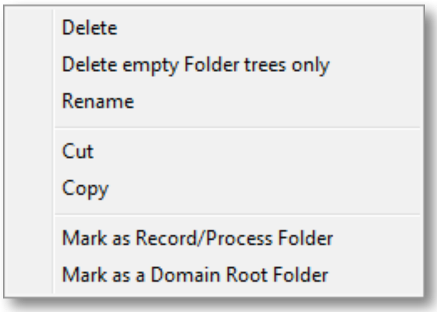
Copy Active files to this folder

This 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 (i.e. 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.

Move Active files to this folder

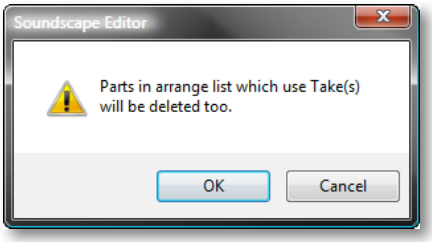
Selecting this option will move the files used be the active arrangement to the currently opened folder. This an easy way to gather all the audio and automation takes and the currently active mix and arr files into a single folder.

Right click on a folder
Right clicking on a folder opens a different context sensitive menu.

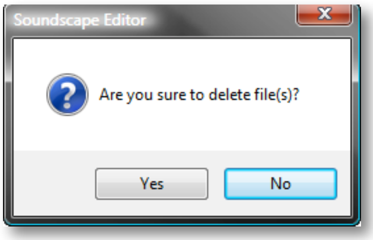


Rename, Cut and Copy work in the same way as in Windows Explorer.
The above described functions to mark a folder are also present in this menu.

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 dialog box will be displayed.



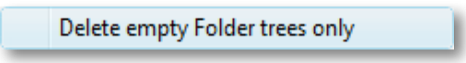
Click on Yes only if you want to continue with deleting the Take(s). If any Part(s) in the current Arrangement reference any Take (s), a further dialog box appears that warns you that this (these) Part(s) will be deleted.



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.

Warning: Remember??? — Backups!!!



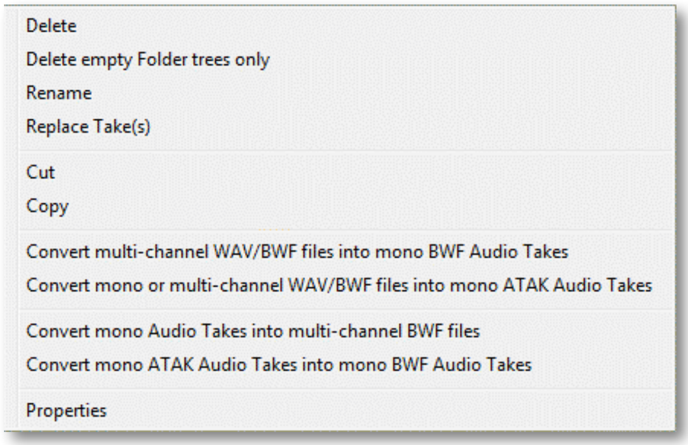
By selecting this item any empty folders / subfolders within the currently selected folder are deleted.

Right clicking on a Take File

Right clicking on an audio Take (ATAK, WAV or BWF) file opens the following menu.

Rename, Cut and Copy behave in the same way as you would expect in Windows Explorer.

Delete empty Folders trees only are also present and have been explained above.



Replace Take(s)

With this 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 menu: Settings|Preferences|Arrangement.

Convert multi-channel WAV/BWF files into mono BWF Audio Takes

This command 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.

Convert mono or multi-channel WAV/BWF files into mono ATAK Audio Takes

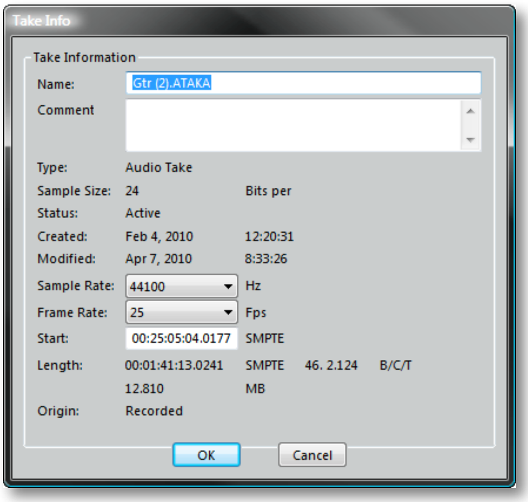
Clicking on this item will convert mono or multi channel (stereo or more) WAV/BWF files into single/multiple mono Soundscape ATAK files, that can directly used inside the Arrange Window.

Convert mono Audio Takes into multi-channel BWF files

Selecting this option 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.

Convert mono ATAK Audio Takes into mono BWF Audio Takes

Selecting this option will convert proprietary SSL Soundscape ATAK files (always mono) into mono BWF files, you can use in any other Audio Application.



Audio Takes

To view Take information, right click on a Take in File Manager and select **Properties**. The dialog box shown to the right will be displayed. You can name the Take, add a comment, status, creation date and time, Sample Rate, Frame Rate, start time, length, size, Sample Resolution (Sample Size), and origin.

Automation Takes

The dialog box includes “Knobs” and “Events” information, and no Sample Resolution.

Auditioning Files and Arrangements

Auditioning (Preview, Playback) an Audio Take inside File Manager is done in the following way:

- 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 Ear Icon (Solo-Tool).
The audio will always be played from the beginning of the Take, including any initial silence (or Amp Hum;-).

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 menu: Settings|Muted Parts/Takes Solo Output|Audio.

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 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.
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 Multi-Channel audio Takes are automatically dragged as a pair or as multiple channels and become Stereo/Multi Linked Parts inside the Arrangement.
This requires that Takes have the same length and the same name (name complemented by an L / R or 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 menu: 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 (menu: 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 (i.e., 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)

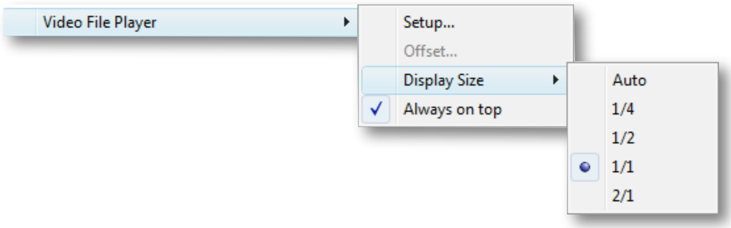
Dragging Takes or Arrangements to a Locator using the [Alt] Key
To drag a Take or Arrangement from the File Manager to a Locator position in the Arrange window, press the [Alt] key while dragging/dropping and the beginning or end of the Part or any Part inside the dragged Arrangement will be positioned at the nearest Locator automatically.

Video File Player Window

Soundscape V6 has a built-in Video Player that plays back AVI encapsulated Video Files synchronised to the Soundscape Timeline sample accurately.

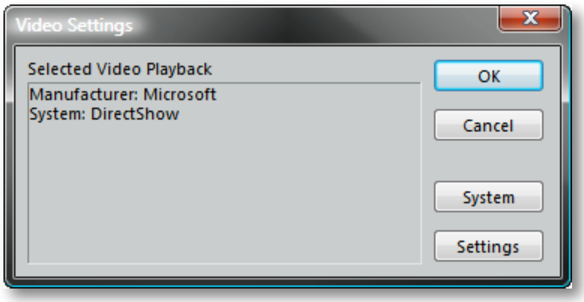
You can open Video Files for synchronised Playback with Soundscape in the **File Menu|Open Video File.**
To open or hide the Video Window deselect or select the following entry: menu **View|Video Player** or press the [V] key on your keyboard.

You find the following Video File Player Setup routines in the menu **Settings|Video File Player.**

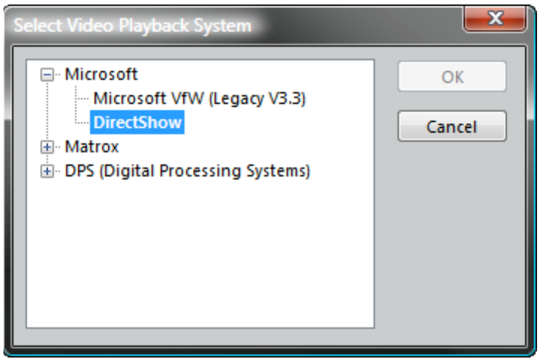


Video System Setup

Clicking Setup opens the Video Settings Dialogue, which shows the currently selected video playback system.



Clicking the **System** button opens a further dialogue where another playback system can be selected.



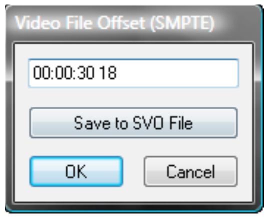
Supported Video Playback systems overview

- **Microsoft Video for Windows (Legacy V3.3):** This legacy video player should be used with old generation video codecs (VfW Codes) or legacy capture cards like the Fast AV Master and Miro DC-30.
- **DirectShow:** Supports different video formats based on Direct Show (DS) software codecs including DV, HDV and others. The Direct Show player can be used with a dual head card, where the video is displayed as a preview in the Video Window and full screen on the second monitor.
- **DPS Reality and Matrox Digisuite:** Supports Playback through legacy DPS and Matrox Broadcast Cards.

Clicking the **Settings** in the previous Video Settings Dialogue button opens the **Frame Delay** setup, where a Video Processing delay can be set from 1 to 20 Frames. This is necessary to compensate for delays Graphics or Video Cards introduce to push data through Hardware Codecs to the Video Out.
Soundscape V6 will effectively play the Video early by the amount of frames that have been set.

Offset

Clicking **Offset** in the submenu displays the Video File Offset window.



The easiest way to set the Video File Offset is to make sure that the SMPTE Offset in the Settings menu is set to 00:00:00:00. This will ensure that the first frame in the AVI file will be displayed. If you have captured with a Time Code superimposed, you can simply read this from the video window and type this in as the Video File Offset. If not, you may have to identify a significant event in the video that can be related to the audio in some way.

In any case, you must ensure that the SMPTE Format (Frame Rate) matches the Time Code format of the captured video. To check that there is synchronisation from the start to the end of the video file, move the Current Locator and check that the video shows the same Time Code as Soundscape’s SMPTE readout.

The **Save to SVO File** button allows you to save the offset as an .SVO file in the same PC folder and under the same name as the video file itself, so that the next time you load the file, the offset will be automatically set. If it is not possible to write to that location (i.e., if the path is “read only” under Windows), the .SVO file will be stored in the “C:\Soundscape\AVI” folder.

With video files captured at a lower Frame Rate than the current Soundscape SMPTE Format, you will find that as the playback of Soundscape advances, there will be missing frames in the video file, so the same frame is displayed twice. However, there should be no overall synchronisation error between video and audio. If you do find a slippage, then the video file may have been captured with dropped frames.

Display Size

When the mouse pointer is on **Display Size** in the submenu, a further submenu is displayed.

- **Auto:** Sets the Video Window Size automatically, normally Soundscape will try to display the original size in the Video Window, if the Window wouldn't into the currently selected screen resolution the Video Window is resized to 1/2 or 1/4 of the original Size.
- **1/4, 1/2, 1/1 or 2/1:** Will resize the Window to a fixed size of 1/4 to 2 times the original size of the Video file.

Always on top

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 V6 this could be the Mixer or Big Current Time window).

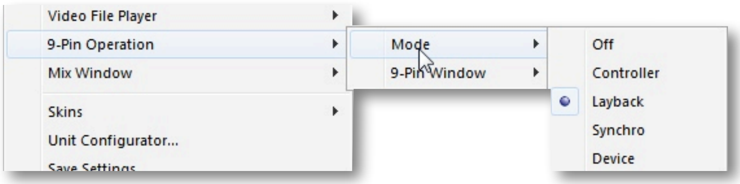
9-Pin Window

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 connected to a 9-Pin device.

The **9-Pin Operation** item in the Settings menu will appear solid if 9-Pin operation is available, or greyed out otherwise.



Clicking on **9-Pin Operation** submenus are displayed that let you choose between Off status and three possible 9-Pin Modes (read below).

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 **Save Settings** under the Settings menu 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 that other window is also set to be “Always on top”: ie. like the Mixer, Big Current Time, and Video File Player windows can also be set this way).

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].

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 the "Set" button 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 the "Set" button is not activated, clicking one of these buttons cues the 9-Pin device to the corresponding "In" or "Out" point.

Preroll

If the "Set" button is activated, clicking this button stores the "seconds:frames" portion of the Current Locator position in the "Preroll" memory of the device.
If the "Set" button is not activated, clicking this button cues the device to the "Preroll" position.

Locator 1-8

If the "Set" button is activated, clicking one of these buttons stores the Current Locator position in the corresponding "Locator" memory.
If the "Set" button 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 analog audio tracks can be armed.
You can specify the number of analog 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 

This 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 record button can only be used if the **Record button Enable** option in menu: Settings|9-Pin Window is selected . If it is not selected the Record 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 V6 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 V6 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 (e.g., 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 V6 as a 9-pin compatible audio device.

All major commands are implemented, e.g., 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 menu: Settings|Master Clock, any received 9-Pin Play/Record start commands will cause Soundscape V6 to start in sync with the next received video frame.

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**
You can open the file with the standard Windows Text Editor.

This line must contain device specific information that is used by Soundscape V6:

Currently the following three values need to be specified per device attached:
- Number of analog audio tracks (0...4).
- Number of digital audio tracks (0...32).

NOTE: A device cannot have both analog 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 analog audio tracks, and its **Edit trigger delay** 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 (e.g., “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.

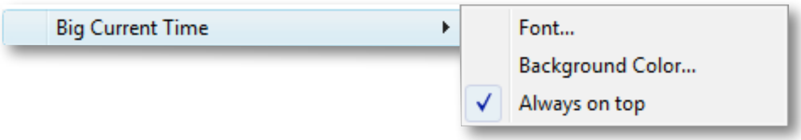
If the [9PinWnd] section does not exist in the INI file yet, then a **Save Settings** operation must be executed first .

Big Current Time Window

The **Big Current Time Window** is 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
(in menu: **Settings**).

It can be shown or by hidden selecting/deselecting Big Current Time in the **View Menu** or by pressing the [T] key.

Customise Big Current Time

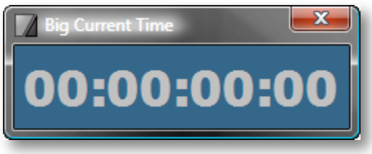


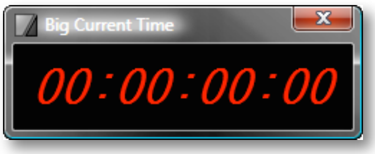
Clicking **Big Current Time** in the Settings Menu opens a submenu with 3 options to alter the appearance and behaviour of the Big Current Time Window.

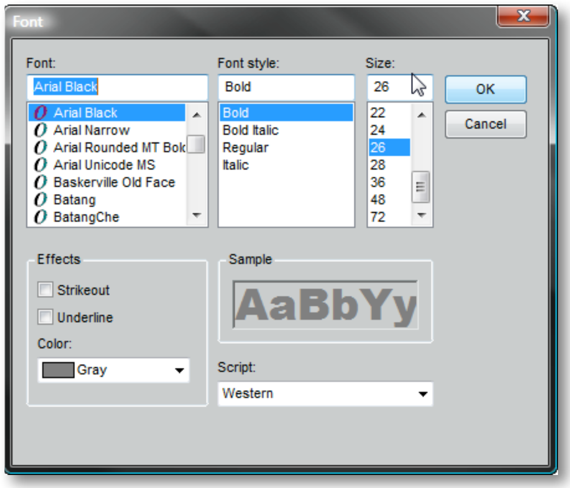
Font

Clicking on Font opens a standard Windows Font selection dialogue, where every installed Font can be selected and both 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 Big Current Time will resize automatically.







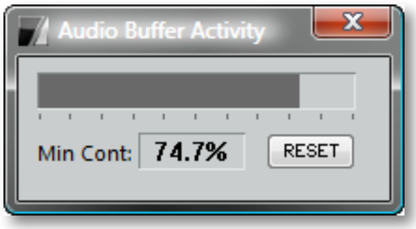
Background Colour

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 the Big Current Time 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 V6, the Mixer and Video File Player windows can also be set this way.

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.

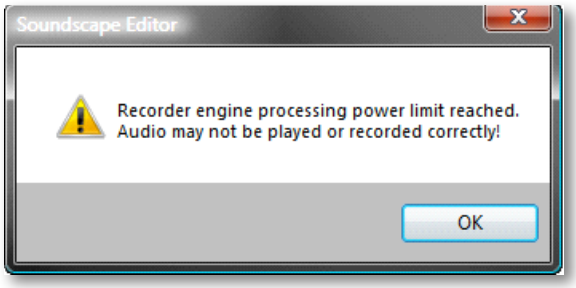
What is a buffer?
The main purpose of Soundscape V6 is to transport Audio Data from Sound to Disk and from Disk to Sound, and in between do some processing with it.

As explained earlier digital systems always need to collect a certain amount of Data first and keep this inside a temporary storage to make sure there is always enough data and the stream doesn't stop (which would in Audio be clicks/drop outs).

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.

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 is the current taxation on the host computer (which an application cannot reliably measure).

If the Buffer Activity drops to 0% you will get the following error:

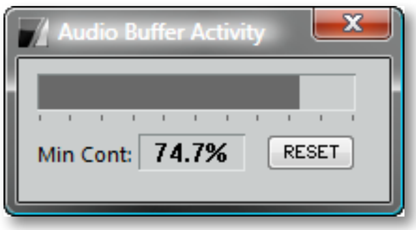


What is measured?
Inside Soundscape V6's revolutionary new Audio Engine any Track is part of a Mini Recorder/Player.
Any Mini Recorder/Player is connected to a DSP on the Audio Card and also to a Hard disk(s).
Any of those Mini Recorders/Players can be located by Windows to a different CPU Core.
128 of those Mini Recorder/Players could work at the same time, so there is some synchronisation and Master Infrastructure that controls and connects to all the Mini Recorder/Players.

All those connections above need to be buffered in order to guarantee a seamless flow of Audio Data at all times.

And these Buffers are 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 is affecting Buffer Health?
Although it has proven to be very difficult to achieve a buffer under run on a modern computer with Soundscape V6 (really!), if the workload of the system becomes high enough and only one 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

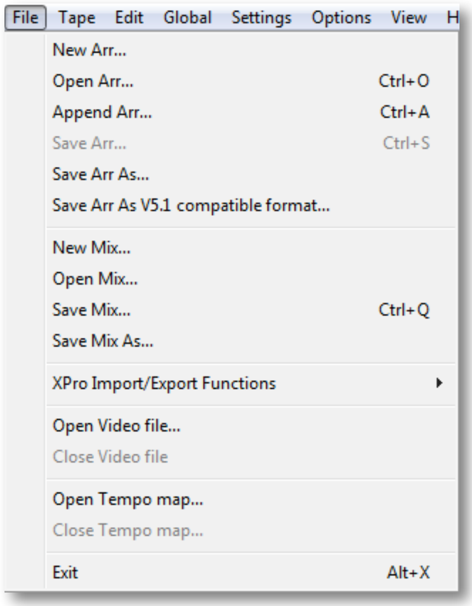
What you can do to avoid Buffer underruns...

- Raise the **Native Mixer Elements Sample Buffer Size** (SettingsIPreferencesIMixer menu) to increase the size of the buffers
- Reduce the amount of random seeking the Disk has to perform, by using the Mixdown tool or Consolidate Process tool to generate new, consecutive Takes.
- Reduce the number of real-time fades by consolidating to a new Take including those Fades
- Use a lower Sample Rate (e.g., 88.2kHz instead of 96kHz).
- 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

The Menu Reference Chapter describes every function and feature in Soundscape V6 in the order they appear inside the Main Menu Structure.
Some Menu's and Menu Entries are only accessible when Soundscape V6 is in Stop Mode. Whenever a Menu is not accessible during Play/Record or certain entries are greyed out/dimmed, please Stop Soundscape V6 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 a certain Point in Time (probably on a rainy Saturday) with the system switched on.

The File Manager Window, however, completely relies 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 options to import or export to MP3, Pro Tools, SDR and OMF formats, open or close Video files and tempo maps.

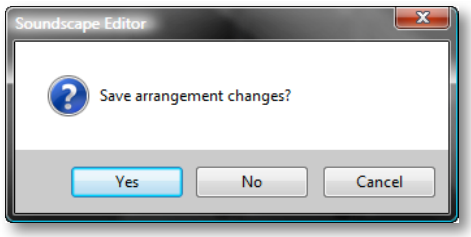
Finally, Soundscape V6 can be closed from this menu.

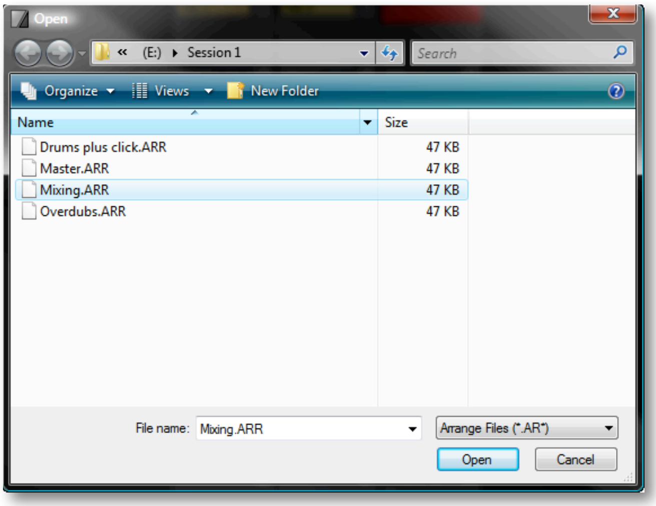
The File Menu can not be accessed while Soundscape V6 is in Play or Record Mode.

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 following dialog box will appear, giving you the option to save it first:

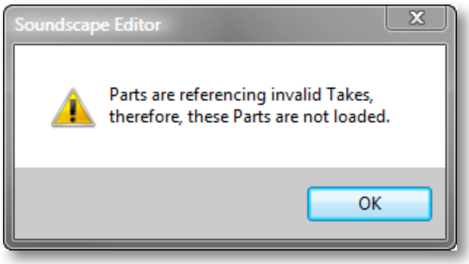




Open Arr... invokes a standard Windows dialogue box:

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, the following warning will be displayed:



Any Parts that reference Takes that are not present will be removed from the loaded Arrangement. 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.

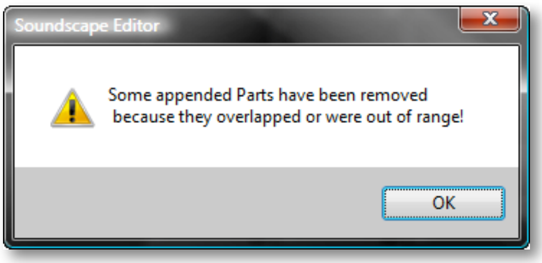
- Otherwise the Arrangement will still play correctly except for the missing Parts, but if you save it under the same name the missing Parts will not be referenced anymore.

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** (menu: 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.

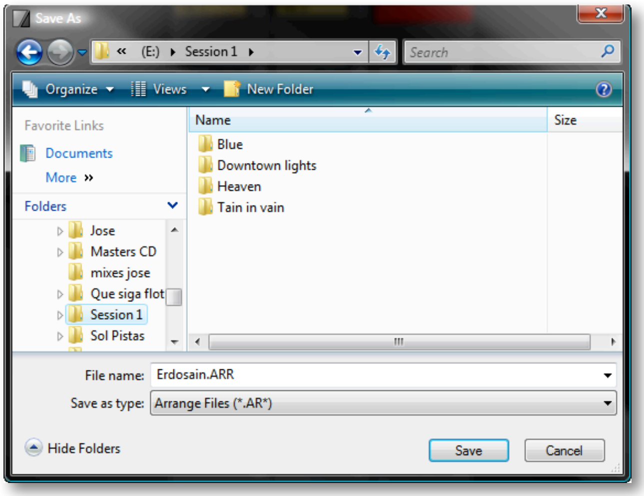
Parts which would be out of range are not loaded either. In such cases the following warning will be displayed:



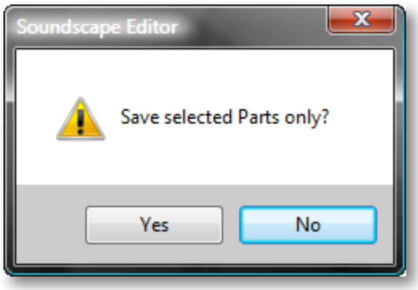
All the parameters saved within an Arrangement file will be loaded from the appended Arrangement, including Sample Rate, Time Code format, SMPTE offset, Varispeed setting, Record setup, Automation setup, Tempo, and Time Signature.
Therefore, if you wish to retain the settings of the current Arrangement, 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 Windows file Browser will open:



Saving selected Parts as an Arrangement
If any Part is selected when you **Save Arr as**, a dialogue box will be displayed:

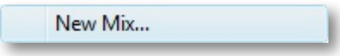


If you click **No**, the whole Arrangement will be saved, as described above.

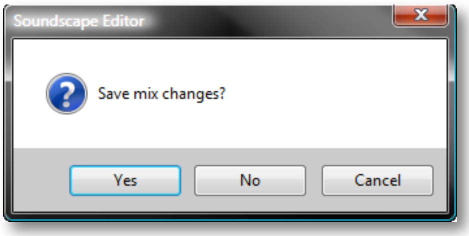
If you click **Yes**, only the selected Parts will be saved, with their individual parameters such as time position and virtual track position in the Arrangement, and track assignment, but without all the usual Arrangement parameters listed above.

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.

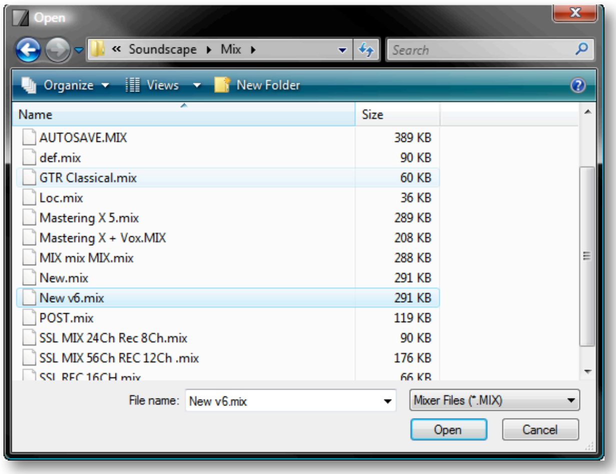
In order to keep compatibility with legacy SSL Soundscape Editor systems (5.1-5.5), you can **Save Arr As V5.1 compatible format.** This option is only available when no WAV/BWF Takes are referenced by any Parts inside the Arrangement (in other words only the Soundscape .ATAK Format is used).



Clicking **New Mix** will create a new, empty Mixer.
If a Mixer is already active and has been edited since the last save, the following dialog box will appear:



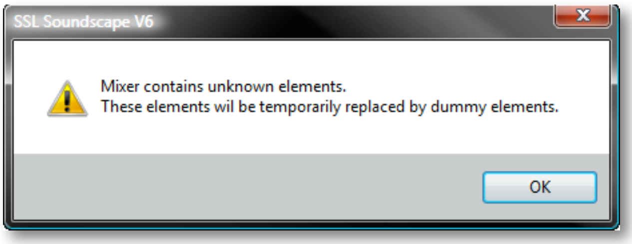
Open Mix opens a standard Windows file browser to search and load an SSL Mix file from any logical PC drive or network location:



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 are attempting to load Mixers on a different SSL system containing different hardware or software configurations (e.g., different number of hardware units or plug-ins that are not available).

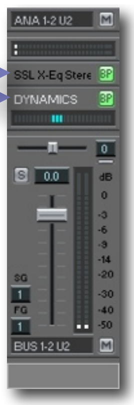
When the Mixer you are opening includes one or more mixer elements not available on the system, the following warning will be displayed:



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.

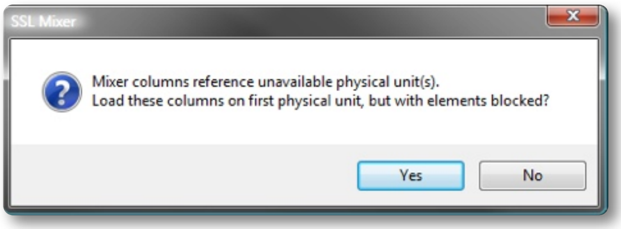
When the same Mixer was loaded onto a system where these plug-ins are not available, the Mixer shows the missing plug-ins as red dummy elements:

Original Mixer with Duende's VST X-EQ and SSL Audio Toolbox Dynamics plug-ins.



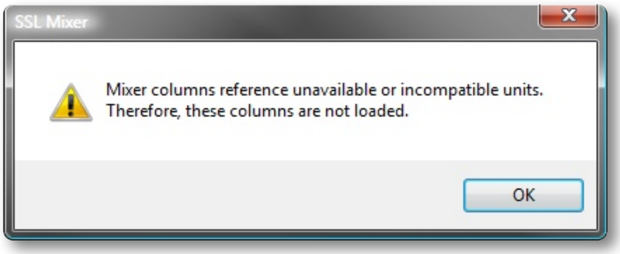
The same Mixer, running on a different system, with both plug-ins replaced by dummy mixer elements.

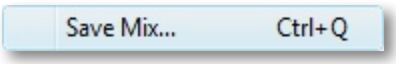
If the Mixer you are opening contains columns which refer to SSL physical units (i.e., hardware units) that are not available, the following dialogue box will let you decide to if these columns should be loaded with their elements blocked (i.e., muted) or skipped:



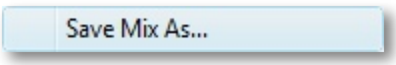
Clicking **Yes** will load these mixer columns. The Mute tool can be used to unmute them.
These columns will be assigned to the highest numbered unit within the system. This may cause mixer elements to become muted in other mixer columns.

Clicking **No** will not load these mixer columns. The following warning will be displayed before the Mixer is loaded:

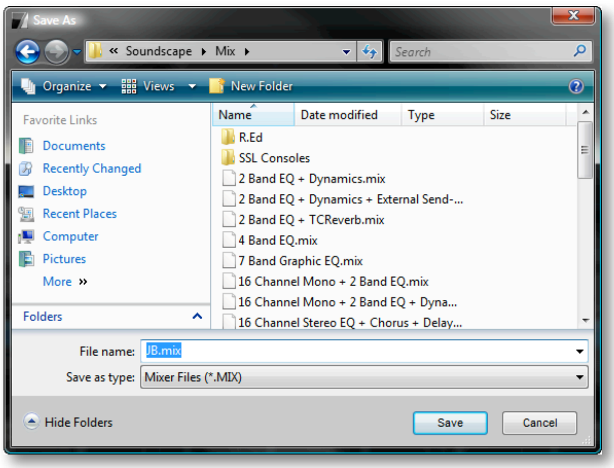




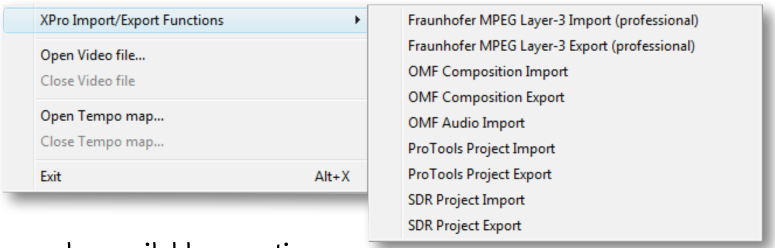
Save Mix... saves or updates the active Mixer. If the active Mixer has not been saved previously, a standard Windows "Save As" browser dialogue opens.



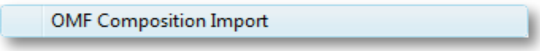
Save As opens a standard Windows browser dialogue. The Mixer needs to be named or renamed and you can browse your local and network drives to select the destination path.



XPro Import/Export components 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.



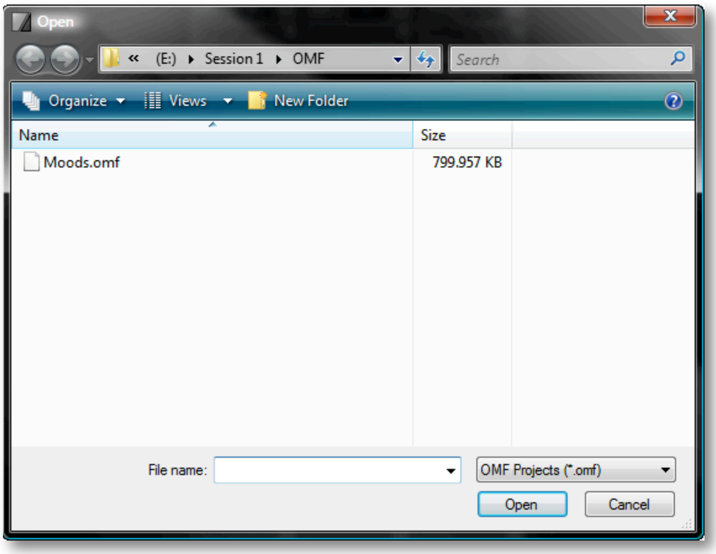
OMF Composition Import/Export and OMF Audio Import are also available as options, for which the software must be installed separately.
All available XPro component options are organised in a submenu.



The OMF Composition Import and Export option and the OMF Audio Import option are available if you have installed the OMF Import/Export optional Plug-In Module. A password must be entered under the Options menu.

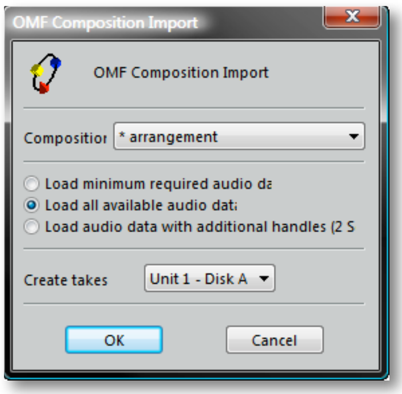
After selecting the **OMF Composition Import** a standard Windows file browse dialogue appears to locate and select an OMF Composition stored on any logical PC drive.

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**:

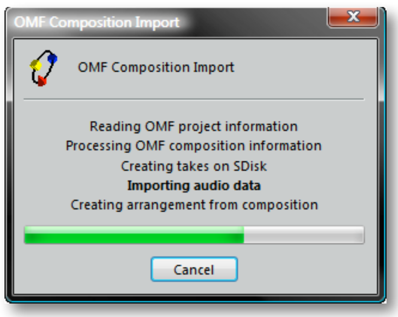


After selection, clicking **Open** will open a dialog box with audio import options:

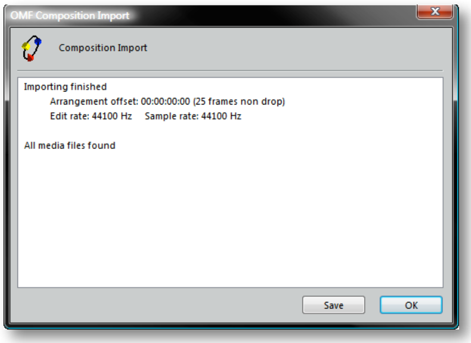
- **Load minimum required audio data** imports only the audio used by the Parts
- **Load all available audio data** imports the whole Takes even if the Parts are shorter
- **Load audio data with additional handles** imports the audio used by the Parts plus handles.



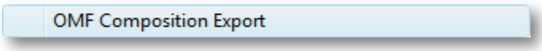
Clicking **OK** will start the import and a progress Window will be displayed:



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.



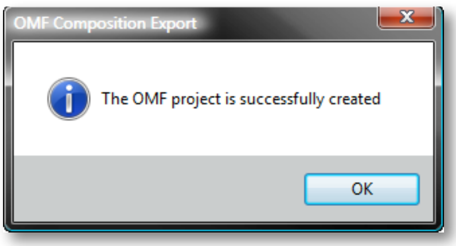
Click on **OMF Composition Export** to open the OMF Export options:

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.

Save will initiate the export and a progress Window will be displayed...

...followed by a Success Message when the Export is finished.

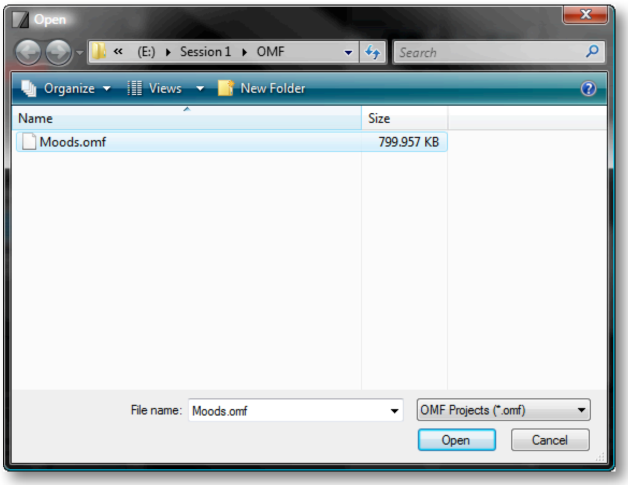


NOTE: The Export tool cannot handle Arrangements which contain Part overlaps or crossfades, therefore it might be necessary to use the **Flatten Arrangement** [Ctrl]+[Shift]+[K] command in the Global Menu.

OMF Audio Import

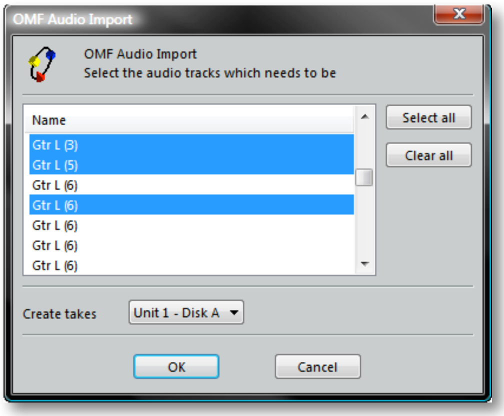
The **OMF Audio Import** option lets you import the individual audio files from an OMF composition as Takes. These files are not loaded as Parts in the Arrangement but created as new Takes in the Recording/Process Folder.

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**.

You can 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...

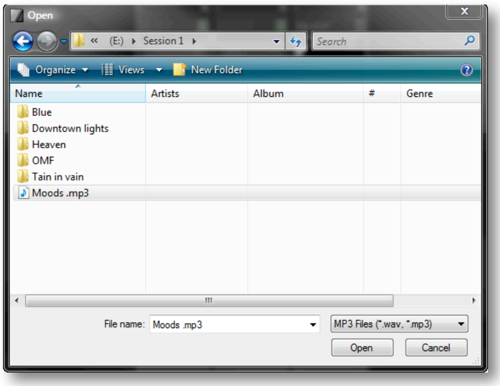
...and a progress Window will be displayed

The imported files will be created as Takes on the active Record/Process folder.

Fraunhofer MPEG Layer-3 Import (professional)

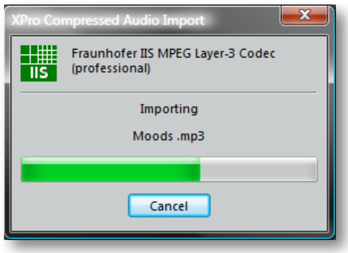
IMPORTANT: The Fraunhofer MPEG Layer-3 Import and Export option are only available if you have installed at least Windows Media Player 10.
If you only see the Import Option and it is without the (professional) extension, please first install a new Version of Windows Media Player (V10 or above) and then re-Install your Soundscape V6 software (nothing will be lost!) to activate the MP3 Codec in Soundscape V6.

Locate and select any MP3 file stored on any PC drive for import as a Soundscape audio Take. 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.**

A Progress Indicator appears.



When the Import is done, new Takes have been created in your current Record/Process Folder.

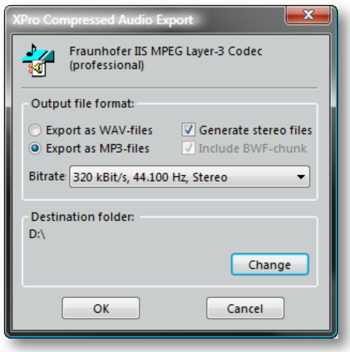
Fraunhofer MPEG Layer-3 Export (professional)

Selecting this option either **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.

Click **OK.**

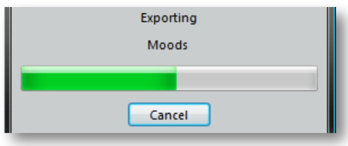
In the next Dialogue the Output Format needs to be selected:

- **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 to Wav File 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 Project Import

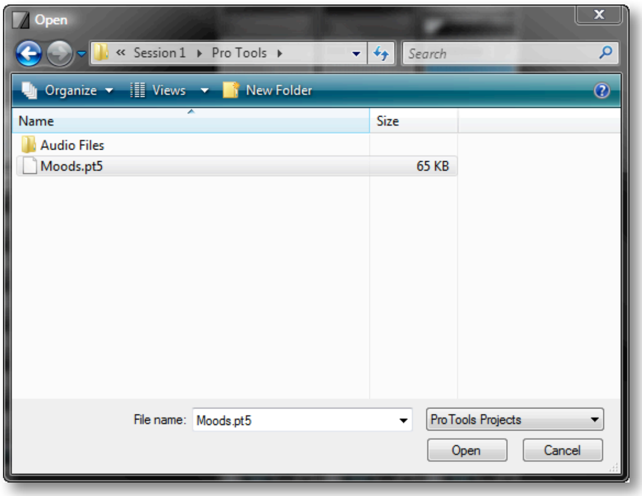
The **ProTools Project Import Function** enables Soundscape to read AVID ProTools© Projects, created with earlier Versions of this DAW Software (V3.-V5.x).

IMPORTANT: If you need to exchange ProTools Projects in a later Format (6,7,8.x) you can purchase SSL Pro-Convert, which translates Soundscape Arrangements into the latest ProTools Session formats and vice versa. For more Information about SSL Pro-Convert, please visit the Product Website at: **http://www.solidstatelogic.com/music/pro-convert/index.asp**

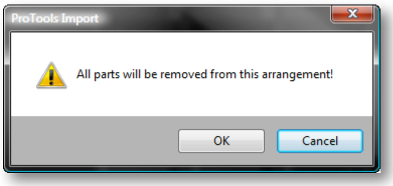
NOTE: There is no need for a password to authorise the ProTools Import/Export function, even though there is a line for entering one in the Optional Modules dialog box .

Click on **ProTools Project Import** and browse to the Session you want to import.

Select the Project and click **OK.**

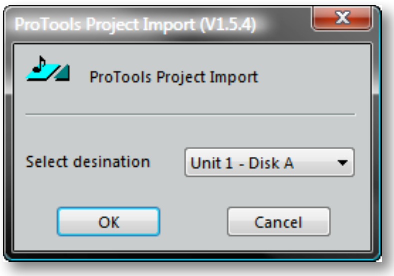


If there are any Parts in the Arrange window, a message box will warn you that these Parts will be removed from the Arrangement. Click **Cancel** if you need to save the current Arrangement before proceeding, otherwise click **OK**.

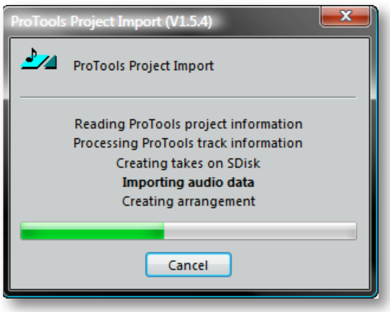


In Soundscape V6 you don't have to select a Destination, the Project Import will go into the currently selected **Record/Process Folder.**

Click **OK.**



A Progress Window will appear until the Import is done.



The audio files from the 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 the **Save Session Copy In** item (File Menu) to create a copy of the Project in a new folder.
In the dialog 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, FAT-32, 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).

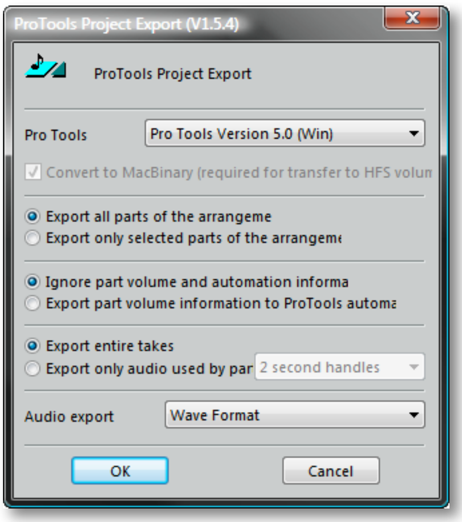
- In case the files are stored on a HFS(+) volume, conversion to MacBinary format is required for SDII and V3.x, V4.x and V5.0 Mac-version project files to get single files that contain both data and resource fork.
- If the projects are stored on a FAT-32 volume written by a Mac operating system, no 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 ProTools 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 ProTools Session formats and vice versa. For more Information about SSL Pro-Convert, please visit the Product Website at: **http://www.solidstatelogic.com/music/pro-convert/index.asp**

ProTools Project Export

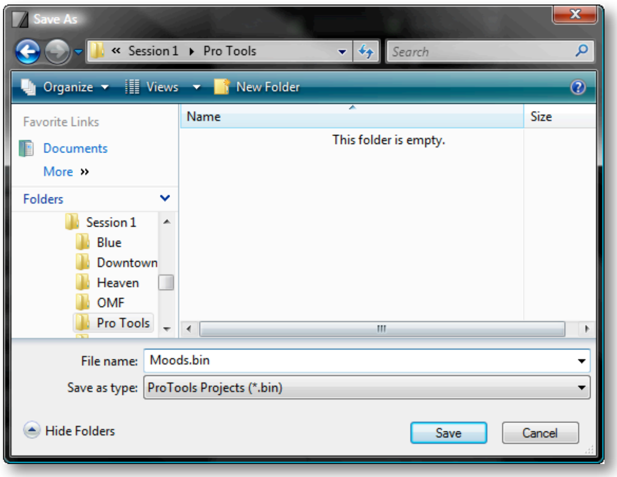


ProTools Project Export saves the current Arrangement as a legacy ProTools Session.

- **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.

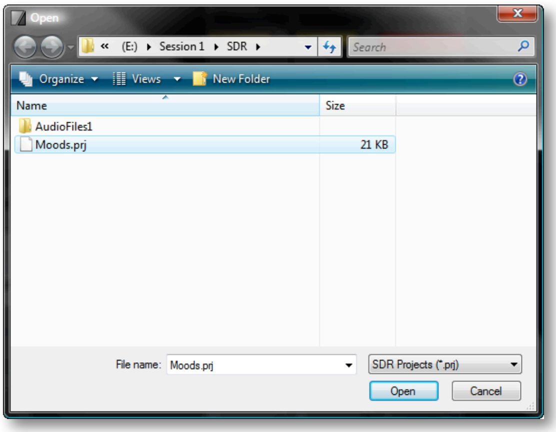
Additional information regarding Pro Tools Export

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.

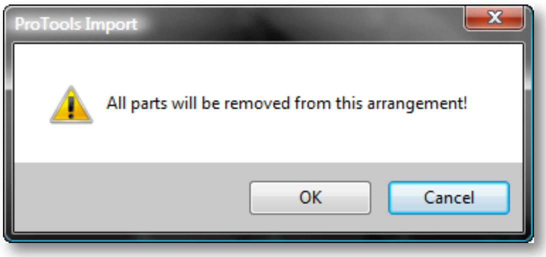
IMPORTANT: If you need to exchange ProTools Projects in a later Format (6,7,8.x) and also Export Volume/Pan Automation, Markers and Crossfades, you can purchase SSL Pro-Convert, which translates Soundscape Arrangements into the latest ProTools Session formats and vice versa. For more Information about SSL Pro-Convert, please visit the Product Website at: **http://www.solidstatelogic.com/music/pro-convert/index.asp**

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.



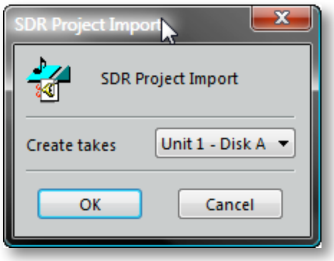
Click **Open.**

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



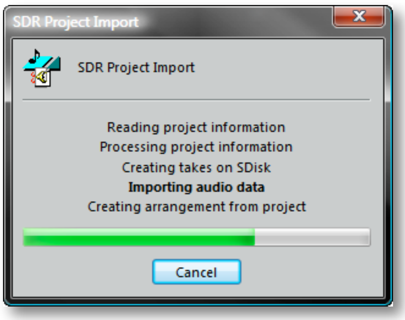
Click **OK.**

In Soundscape V6 you don't have to select a Destination, the Project Import will go into the currently selected **Record/Process Folder.**



Click **OK.**

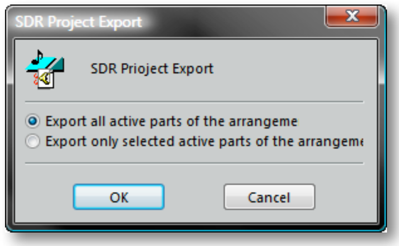
A Progress Window will appear until the Import is done.



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.

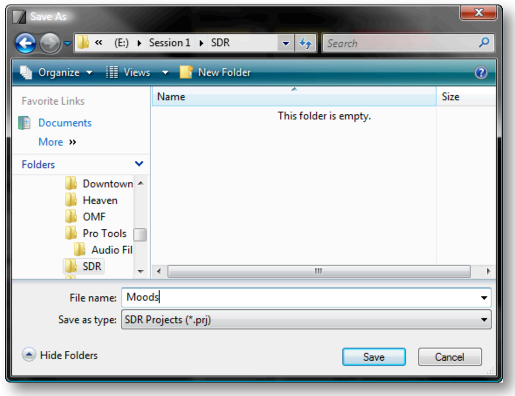
SDR Project Export

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 and give the Session a name...



Click **Save**.

The Export will start and a Progress Window appears until the export is finished.

Video Files

Open Video file...

Open Video File allows to you select a video file stored on the PC for playback in Soundscape V6's Video File Player. By default the file will automatically play from the start of the current Arrangement, however an offset can be entered (menu: Settings| Video File Player|Offset). 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.

Close Video file

Close Video File will stop the Video File Player Module inside Soundscape V6. Closing/Hiding the Video File Player Window leaves the Video Playback active but invisible.

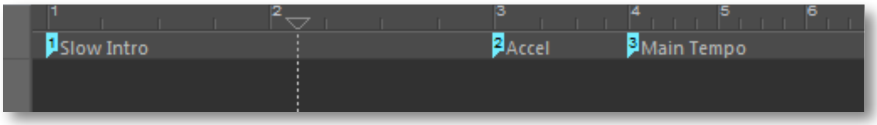
Tempo Maps

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 V6, 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 .

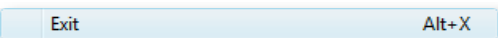
Also if you are working with Audio files that have been created with a Midi Sequencer (ie. Track Bounces of Virtual Instrument Tracks) using dynamic Tempo changes, the imported Tempo Map allows you to still edit with Snap in a musical grid.

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.

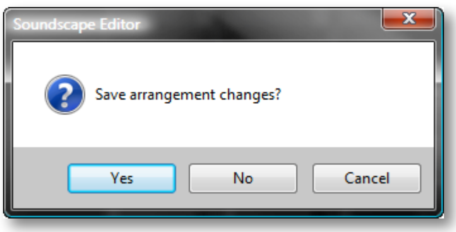
IMPORTANT: Some sequencers provide a tempo map that has a slightly lower resolution than their internal tempo map. In such cases the midi file used to load the tempo map into Soundscape V6 should be imported back into the sequencer as well.

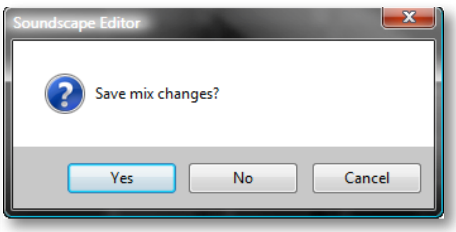
Clicking this item will close the tempo map and Soundscape V6 will return to a static Tempo and Sign, defined by the Tempo and Sign settings.

Exit



Clicking Exit will initiate the SSL Soundscape V6 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 (i.e., not to close the program):





Tape Menu

The Tape menu is divided into four sections.

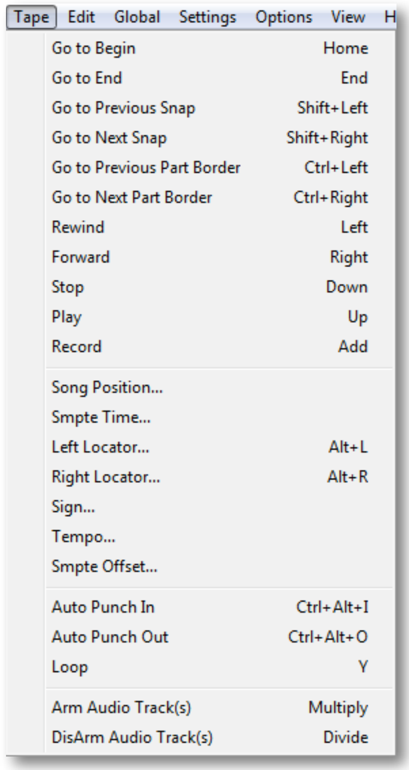
The items in the top section duplicate the function of the corresponding transport bar buttons.

Clicking on any of the items in the second section of the menu will open a dialog 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.

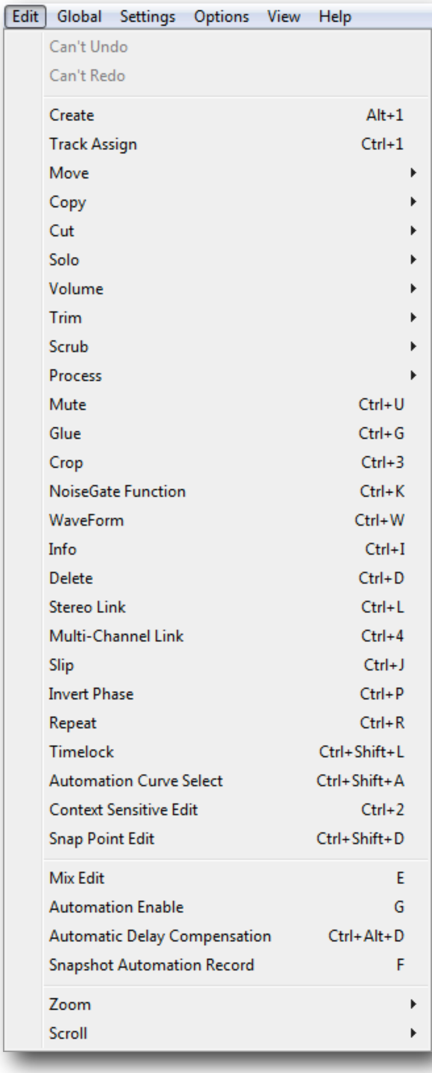
The items in the lower section of the menu can be used to arm or disarm all the existing audio record tracks.

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 menu: Settings|Preferences|General|Undo/Redo levels.
Undo and Redo can also be performed using the [Ctrl]+[Z] and [Ctrl]+[Y] key combinations respectively.

Saving an Arrangement or closing Soundscape V6 will clear the Undo/Redo Database and "really" perform 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.**

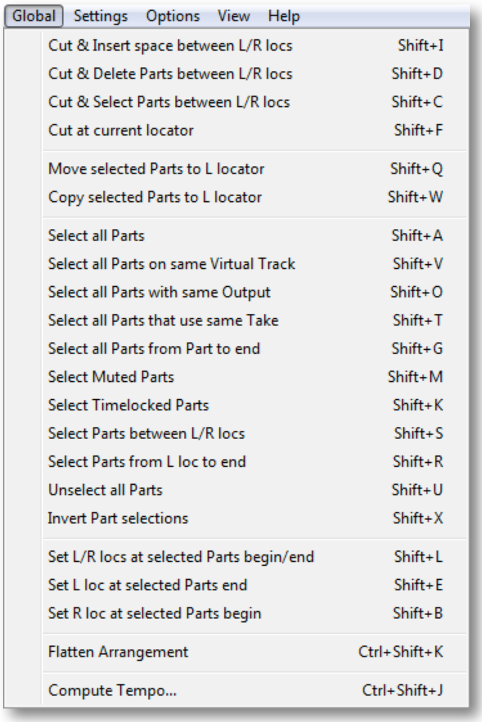
Zoom and Scroll

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).



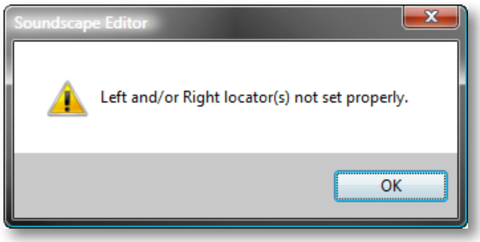
Since all Global functions can **seriously accelerate the editing workflow**, it is a good, if not very good idea, to not only learn how they work but also to **memorise and use the keyboard shortcuts.**

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)

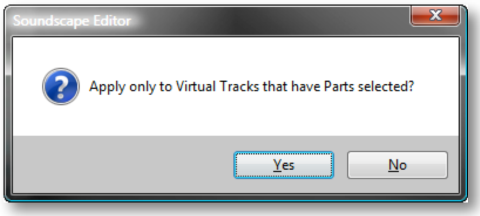
...if the Locators are set incorrectly (i.e. Left Locator is placed **before** the Right Locator), when trying to use the function the following warning will be displayed:

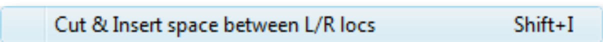


Click **OK**

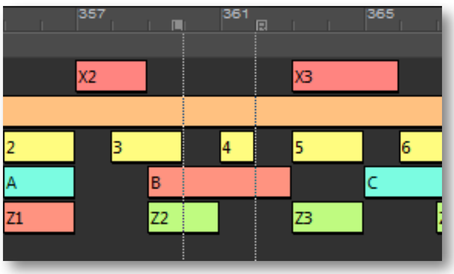
and reposition the Locator(s) and perform the Global Command again.

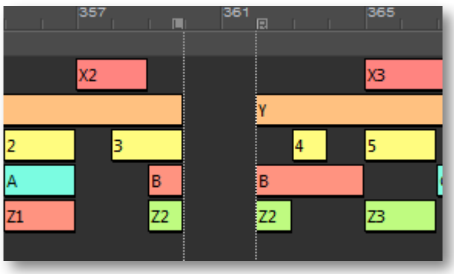
For some of the global functions, if any Part or group of Parts is selected before the edit is made, a dialog box will be displayed, which lets you decide if the edit should be restricted to the virtual track(s) that Part or group of Parts is on.





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 (menu: Settings|Preferences|Arrangement) any Markers after the Left Locator will be moved with the Parts.

Cut & Delete Parts between L/R locs Shift+D

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 (menu: Settings|Preferences|Arrangement) any Markers after the Left Locator will be moved with the Parts.

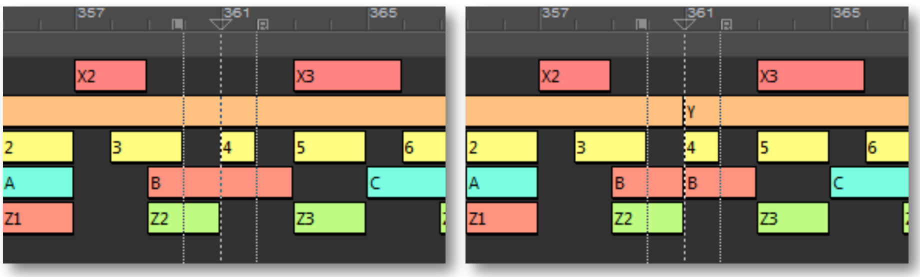
Cut & Select Parts between L/R locs Shift+C

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)

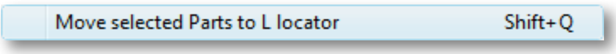
This function can be restricted to certain virtual tracks by selecting at least one Part on each targeted virtual track before performing the edit.

Cut at current locator Shift+F

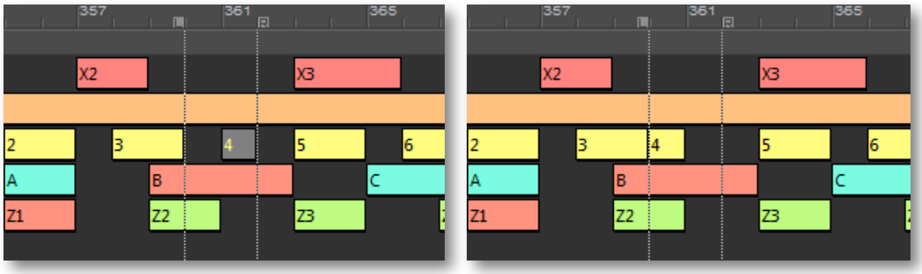
When Cut at current locator is used, a cut is made to all Parts (or selected Parts) at the Current Locator position.

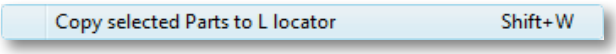


This function can be restricted to certain virtual tracks by selecting at least one Part on each targeted virtual track before performing the edit.

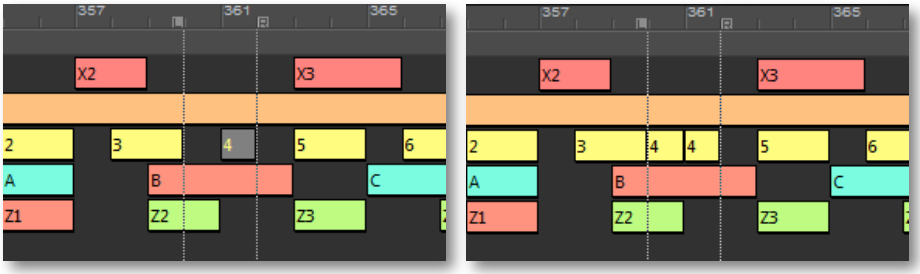


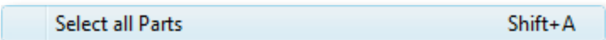
When **Move selected Parts to L locator** is used, any selected Part(s) are moved to the Left Locator.



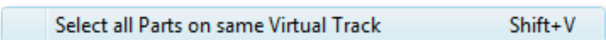


And when **Copy selected Parts to L locator** is used, any selected Part(s) are copied to the Left Locator:

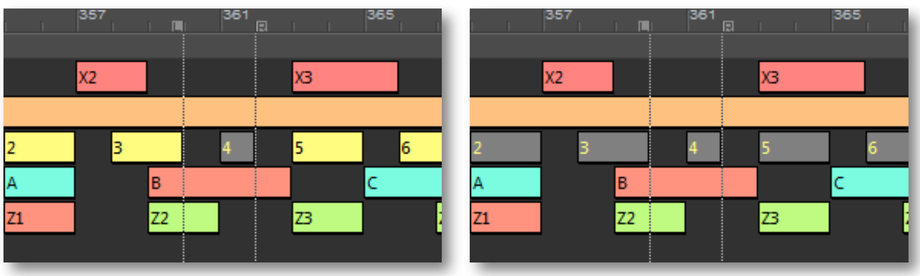


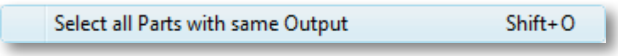


Select all Parts will select any Part in the Arrangement.

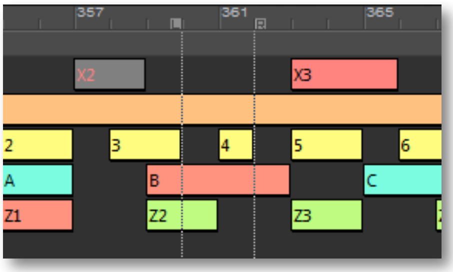


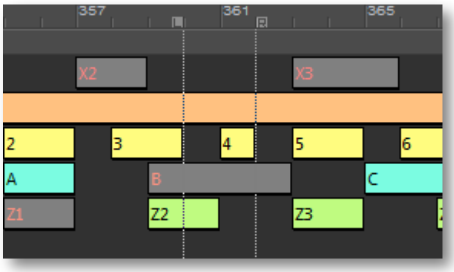
In order to use the **Select all Parts on same Virtual Track** function, at least one Part should be selected. All the Parts that are on the same virtual track as the selected one(s) will be selected:

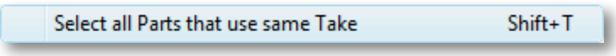




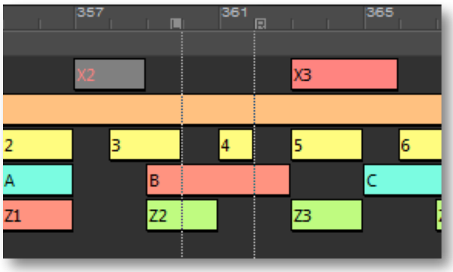
Select all Parts with same Output will select all Parts that have the same track assignment as the selected one(s). At least one Part needs to be selected first.

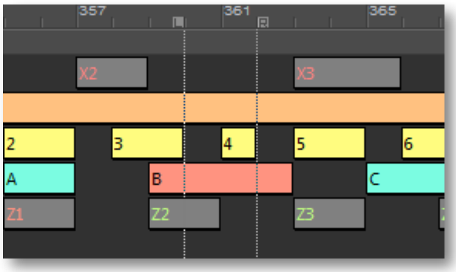


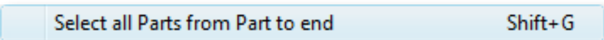




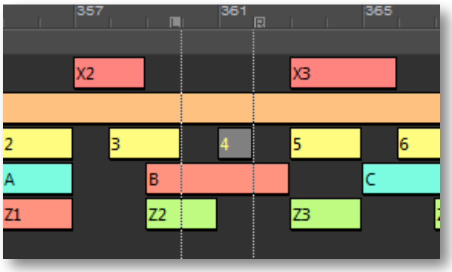
Select all Parts that use same Take function, at least one Part should be selected first. All the Parts that reference the same Take as the selected one(s) will become selected.

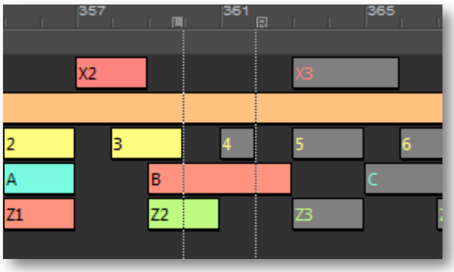






Select all Parts from Part to end [function, at least one Part should be selected first. All the Parts that are positioned to the right of the selected one(s))or that have the same beginning position) will be selected:

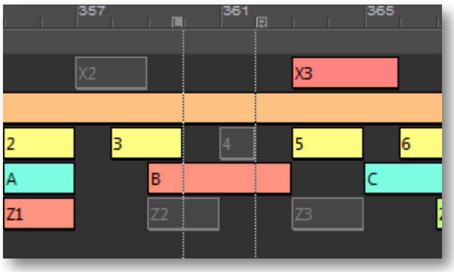
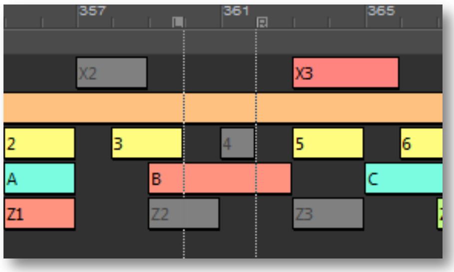




NOTE: This function can also be used by holding [Ctrl]+[Shift] before selecting a Part.

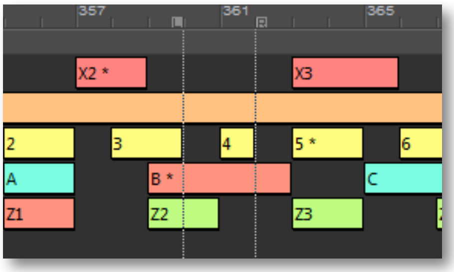
Select Muted Parts   Shift+M

Select Muted Parts will indeed select all the muted Parts.

Select Timelocked Parts   Shift+K

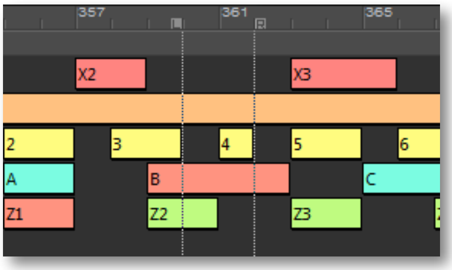
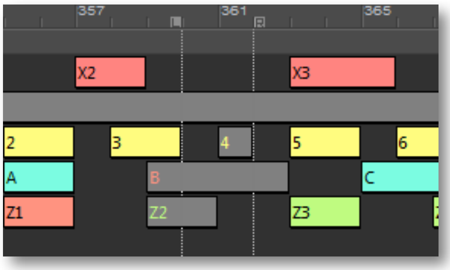
Select Timelocked Parts will select all timelocked Parts (identified by the "*" appended to their name).




NOTE: Please refer to "Timelock tool" of the "Editing Tools" section for a detailed explanation of timelocked Parts.

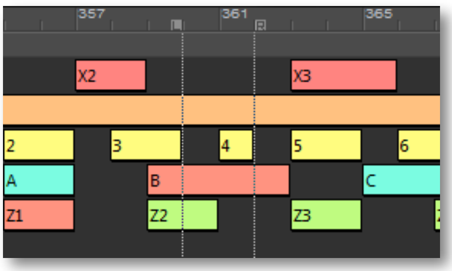
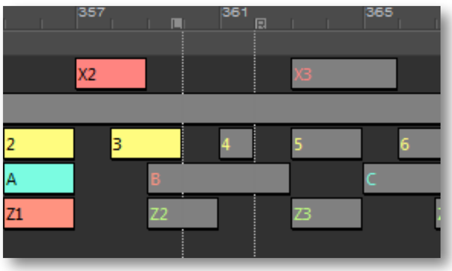
Select Parts between L/R locs   Shift+S

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   Shift+R

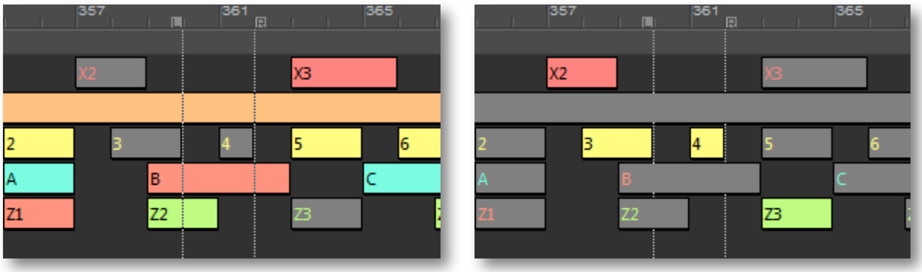
Select Parts from L loc to end will select all Parts of which any section is positioned between the Left Locator and the end.

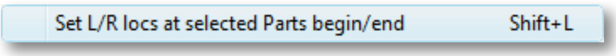



This function can be restricted to certain virtual tracks by selecting at least one Part on each targeted virtual track first.

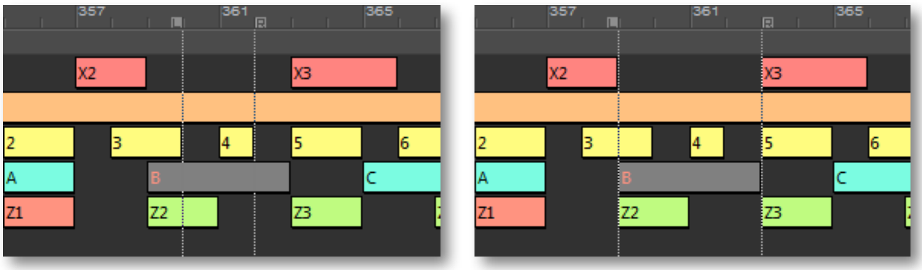
Unselect all Parts cancels all Part selections.

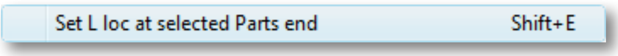
Invert Part selections deselects all selected Parts and unselected Parts are selected instead.



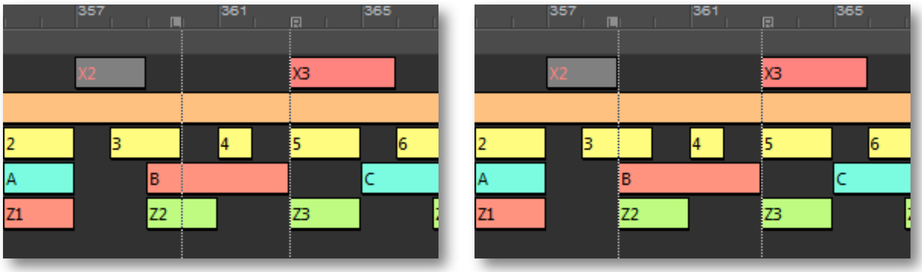


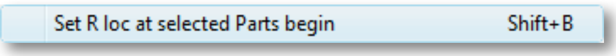
Set L/R locs at selected Parts begin/end positions the Left and Right Locators at the beginning and end respectively of the selected Part or group of Parts. This is especially useful when an area needs to be defined for using the **Mixdown tool** or a loop is selected to define the Tempo with **Compute Tempo** function (see next Page).



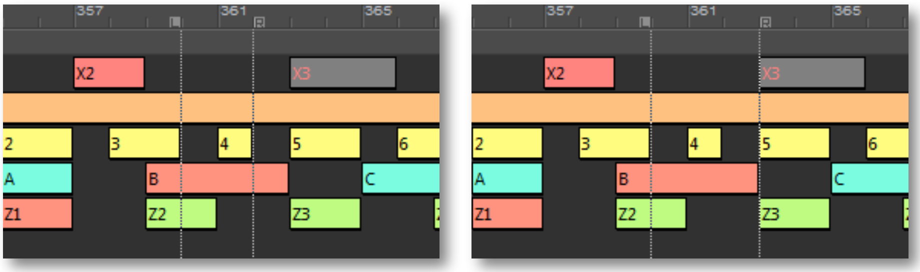


Set L loc at selected Parts end sets the Left Locator to the end of the selected Part (Part X2 in the example below) or group of Parts. This is especially useful when an auto punch in point needs to be set precisely set at the end of a Part.





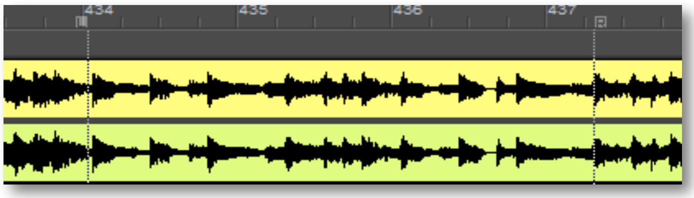
Set R loc at selected Parts begin sets the Right Locator to the beginning of the selected Part (Part X3 in the example below) or group of Parts. This is especially useful when an auto punch out point needs to be set precisely at the beginning of a Part.



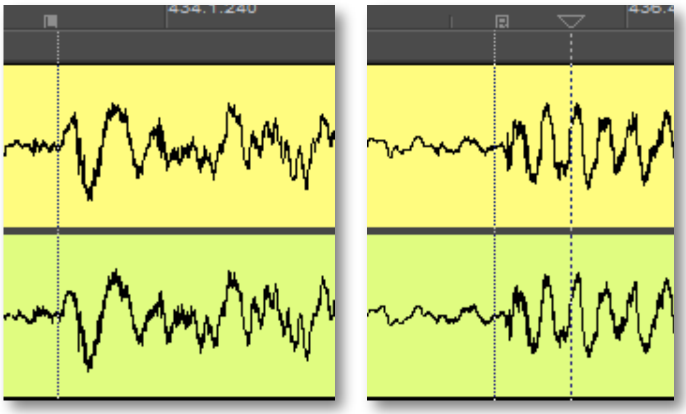
Flatten Arrangement will remove any overlapped Parts, or overlapped sections of Parts and crossfades from the Arrangement . If any Takes become unreferenced as a consequence however, they are not deleted. This is useful for exporting Arrangements to certain formats or with export modules where real time crossfades and overlapping parts are not supported.

Compute Tempo calculates the tempo for Soundscape V6 based on the Left and Right Locator positions, the current Time Signature and the number of bars being considered.

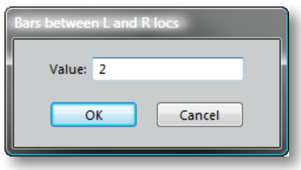
In the following example, it is used to compute the tempo of a drum Part.



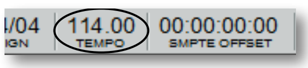
- While the recorded audio Part is played back, drop in the Left and Right Locators on the fly (by pressing the [L] and [R] key) to mark the beginning of bars.
  In our example, the Locators are dropped two bars apart, but any number of bars can be used.



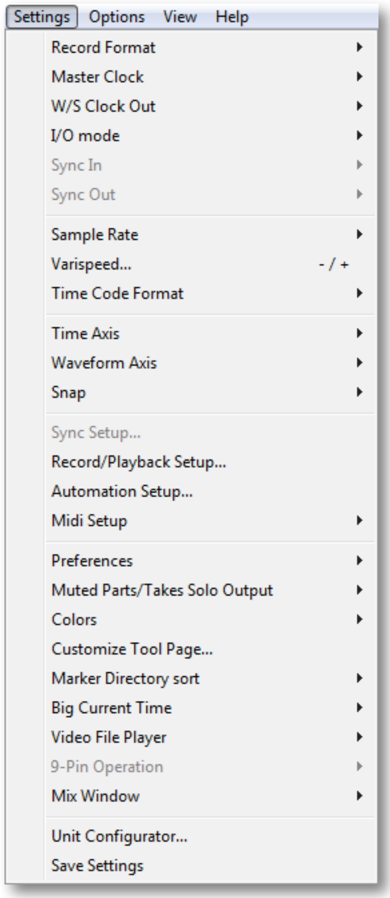
- Stop Playback and zoom in to a sufficient level and clicking on the Left and Right Locator buttons in the Transport Bar (blow the readouts) it is easy to jump between the two marked points.
- It is now possible to place the Left and Right Locators with more precision by visual reference to the waveform. 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 V6 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.

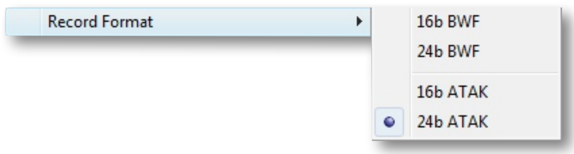
It includes all kinds of settings related to the operation of the hardware and software, display preferences, etc.

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 V6 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.

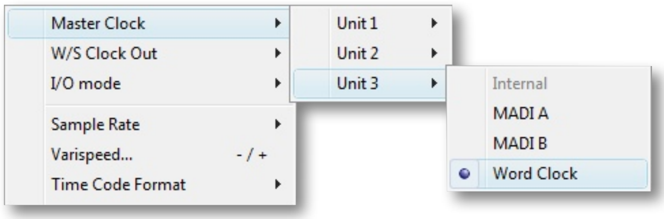
The ATAK format offers an embedded unique ID for every recorded or processed audio file, therefore Soundscape V6 will find the Unique audio files for an Arrangement regardless of their location (within the user definable range of Domain Root Folders).
This makes the ATAK format ideal for projects where 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!

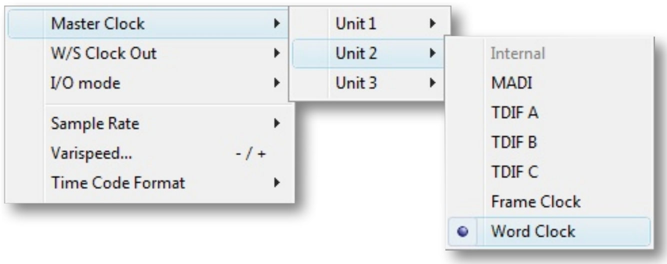
The Master Clock can be altered for each Soundscape Unit separately, that is part of the Unit Configuration.
Available clock sources vary depending on device type (or Alpha-Link interface a Mixpander Card is connected to).

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

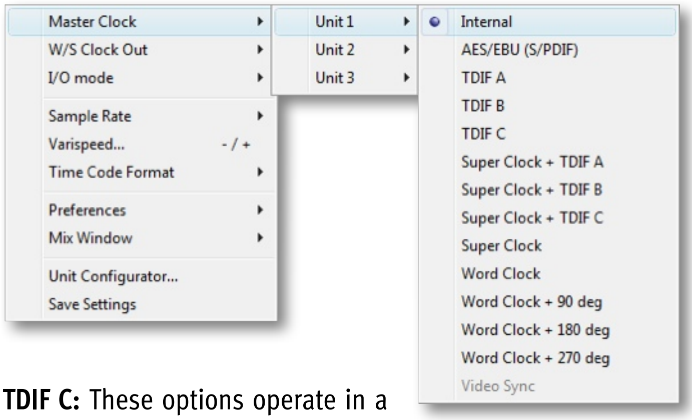


Since Mixpander systems can be connected to various models of the SSL Alpha-Link (or legacy Soundscape iBox Product Range) with several digital I/O options, the available Master Clock options vary (and only depend on the I/O configuration of the connected converter box).

- **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 48kHz. At 96kHz the ADAT Frame can still carry the same amount of data. To achieve the double amount of data per channel an ADAT Frame at 96kHz only contains 4 Channels of Audio at 24Bit (SMUXING). The Clock (or amount of Frames/second) stays at 48000 ADAT Frames/s, therefore the Frame Clock remains 48kHz even when running the System at 96kHz.
- **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

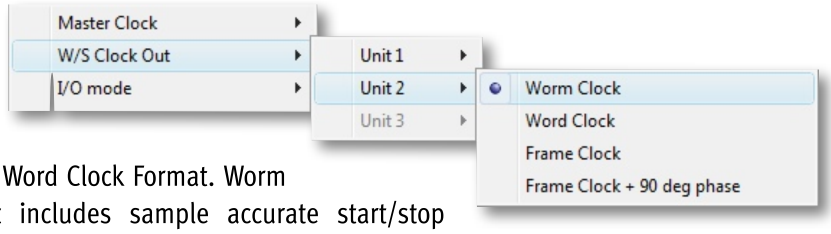
With this menu entry you can control the Clock signal to be transmitted at the Word Clock / Super Clock output connector of the installed device(s). As mentioned above, the available options depend on the installed SSL components.

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.

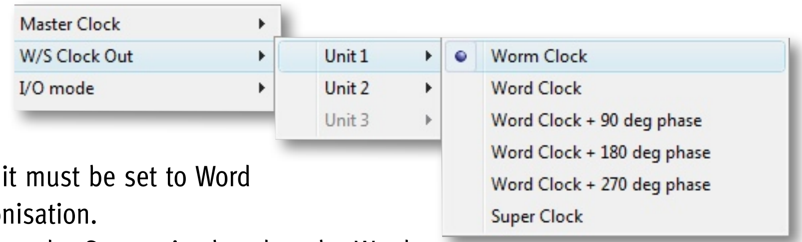
W/S Clock Out settings for a Mixpander and Alpha-Link/iBox units

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)

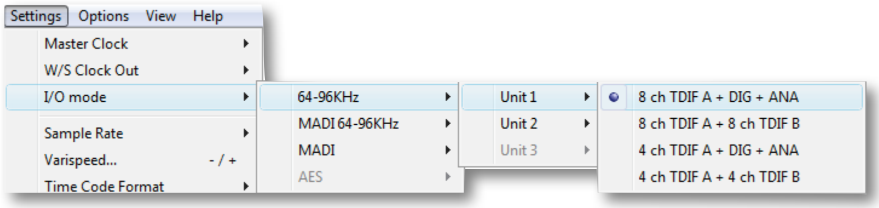
W/S Clock Out settings for a Mixpander connected to a Soundscape 32 unit



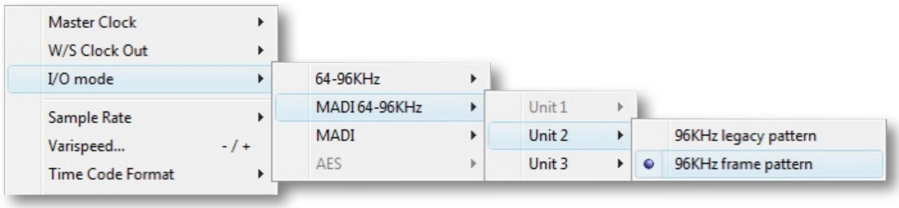
- **Worm Clock:** SSL's proprietary extension of the Word Clock Format. Worm Clock is a modified Word Clock signal that includes sample accurate start/stop synchronisation information for several units. The Master Clock parameter for the slave unit must be set to Word Clock, which also enables Worm Clock 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 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 SMUXed or non-SMUXed Modes for ADAT/TDIF/MADI/AES or different I/O combinations for a Soundscape 32/Mixpander system.

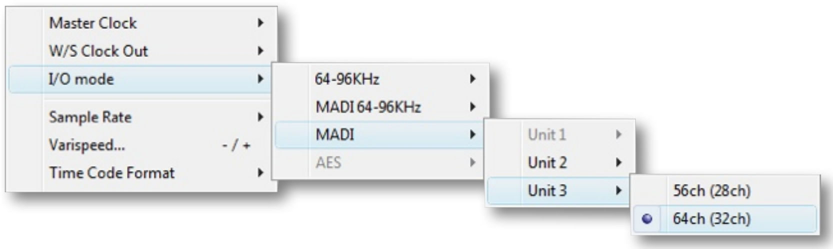


MADI 64-96kHz
In this submenu the MADI Frame Pattern is selected between 96k Legacy Pattern (48K Frame) and 96k Frame Pattern.
The options determine how the 28/32 MADI Channels are encoded inside one MADI Frame.



MADI
This submenu selects the MADI CH Mode (56 or 64 Channels).
In its early days (last Millennium) MADI had 56 CH (at 44.1kHz and 48kHz) 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, the 64 CH 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).

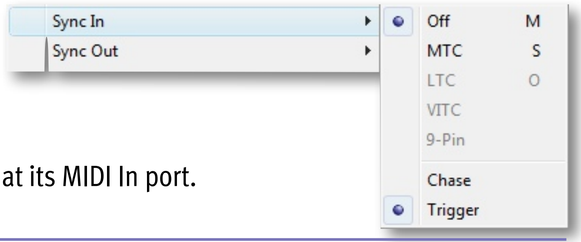


Both standards are equally popular, depending on the variety of applications and industries you work in, you may need to switch between these modes quite often.
At 88.2 kHz and 96 kHz a 56 CH MADI cable operates with 28Ch of audio, while with 64 CH MADI, 32 Channels of audio are transmitted.

This section describes the synchronisation functionality of a Soundscape V6 System that includes a Soundscape 32 Unit connected to a PCI Host IF Card and configured as Unit1.
External Time Code synchronisation is currently not directly available for MX4 and Mixpander units.
However, they support the ASIO Positioning Protocol (APP), allowing APP compatible applications (ie. Steinberg Cubase and Nuendo) to slave to SSL Soundscape.

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.
With the optional Synchronisation Option installed, 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.

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, i.e., if the Play or Record button has been clicked, or if the Auto Play/Stop when slave syncing option is selected under menu: Settings|Sync Setup.

- **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, i.e., if the Play or Record button has been clicked, or if the Auto Play/Stop when slave syncing option is selected under menu: 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, i.e., if the Play or Record button has been clicked, or if the Auto Play/Stop when slave syncing option is selected under menu: 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, i.e., if the Play or Record button has been clicked, or if the Auto Play/Stop when slave syncing option is selected under menu: 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, i.e., if the Play or Record button has been clicked, or if the Auto Play/Stop when slave syncing option is selected under menu: 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 (e.g., 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.

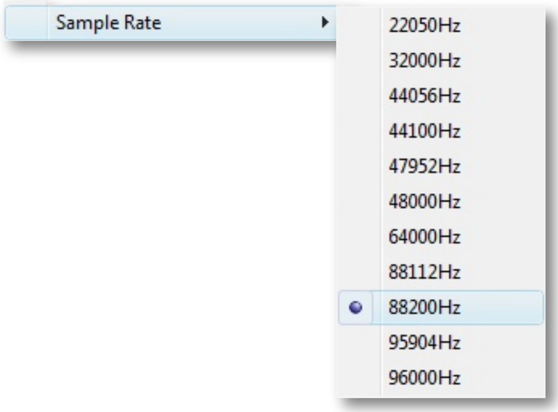
NOTE: Soundscape will only respond to incoming Time Code if it is in Play mode or Record mode, i.e., if the Play or Record button has been clicked, or if the Auto Play/Stop when slave syncing option is selected under menu: Settings|Sync Setup.

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 V6 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.
- **LTC:** This setting should be selected to use SSL Soundscape V6 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 the “Tempo maps” section of the “File Menu” 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

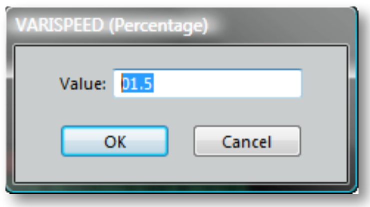
- **22.500Hz** (Multimedia)
- **32.000Hz** (Video and Broadcast Standard)
- **44.056Hz** (NTSC Drop Frame Pull Down of 44.1kHz)
- **44.100Hz** (CD Standard)
- **47.952Hz** (NTSC Drop Frame Pull Down of 48kHz)
- **48.000Hz** (Broadcast Standard)
- **64.000Hz** (Video)
- **88.112Hz** (HD NTSC Drop Frame 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):

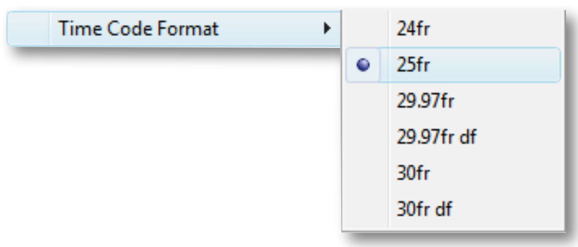


The Varispeed setting can also be adjusted by using the [–] and [+] keys (from the main key area of the computer keyboard only, not from the numeric keypad), which change the varispeed value in –0.1% or +0.1% increments, respectively.

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

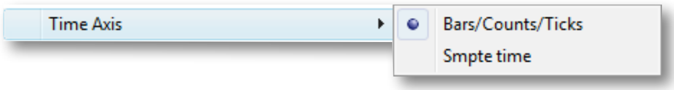


The Time Code Format selects a Frame Rate commonly used as SMPTE or EBU Frame Rates, such as 24 frames per second for film, 25, 29.97, 29.97df, 30 or 30df frames per second for video.
It is required for Synchronisation (external Timecode or APP) and while working with and for Video.

This parameter can also be set in the Status Bar's **FRate** box.

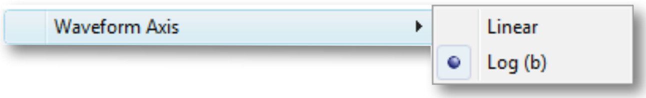
Time Axis

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.



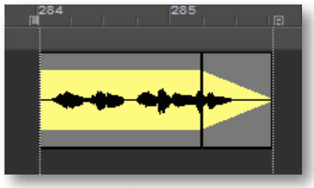
Waveform Axis

This parameter defines the mode in which amplitudes in waveforms, automation data and fade curves will be displayed in the Arrangement.



Compared to the Linear mode, the Logarithmic mode (Log (b)) is visually magnifying lower level details, however leaving less visible space to differentiate higher levels.

Logarithmic Mode will also cause a linear fade to appear curved.



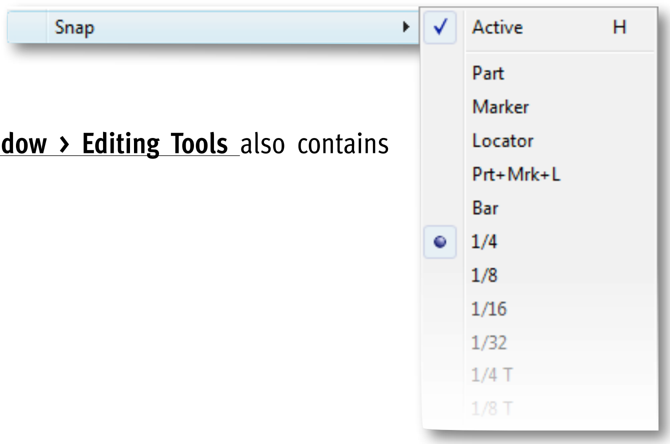
Linear Waveform Axis

Logarithmic Waveform Axis

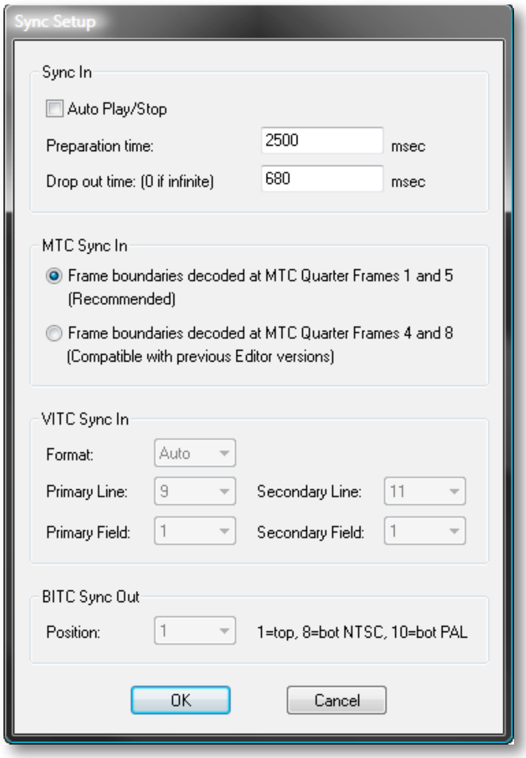
Snap

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.



The Sync Setup window cannot be accessed if the system configuration does not include a Soundscape 32.



Sync In

When the **Auto Play/Stop when slave syncing** box is checked, Soundscape V6 will always respond to incoming Time Code when an external Sync Source is selected (menu: 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 V6 is slaved to external Time Code.

A time value up to 21768ms can be entered for the Sync in **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 (e.g. if the Master stops) this free-wheeling time should be set as low as possible.

MTC Sync In

In the MTC Sync In section, you can 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.

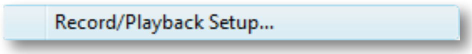
VITC Sync In

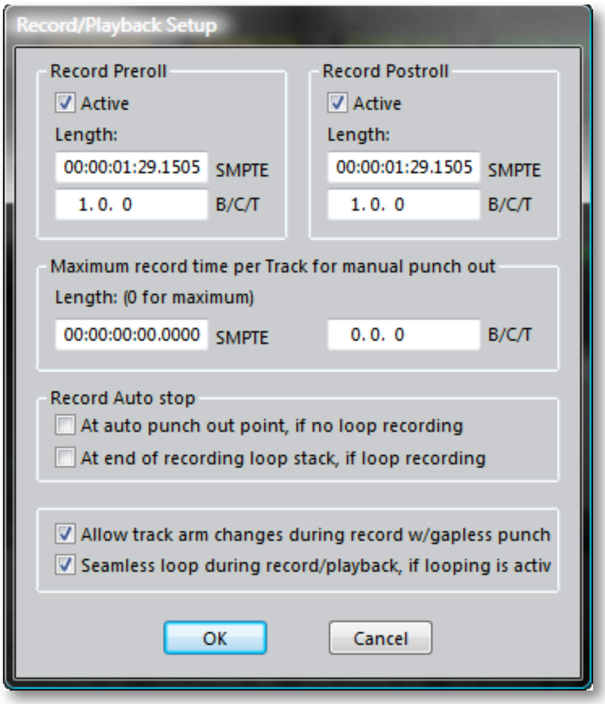
In the VITC Sync In section, the Format box can be used to select a VITC format. The Line and Field settings determine the location in the video signal of the VITC information used to decode the VITC data. If the data in the primary location is corrupt, the secondary location is checked. Clicking the 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.

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 V6 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 V6 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

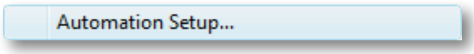
- If **At auto punch out point (if no loop recording)** is checked, Soundscape V6 will automatically stop upon reaching the Right Locator (or the end of the postroll if active) when recording with Auto Punch Out.
- If **At end of recording loop stack (if loop recording)** is checked, Soundscape V6 will 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 V6 needs to record a new signal and play back (muted until punching out) the underlying existing Tracks simultaneously.

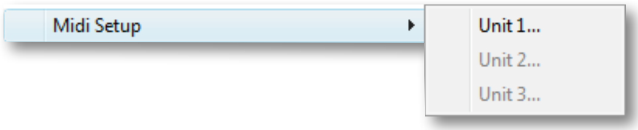
If the **Seamless loop during record/playback, if looping is active** box is checked, audio is played continuously when the Current Locator jumps from the Right Locator to the Left Locator in Loop mode. If the box is not checked, there is a gap in the audio.
Seamless looping is ideal for certain tasks, 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 (e.g., 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.

Automation Setup



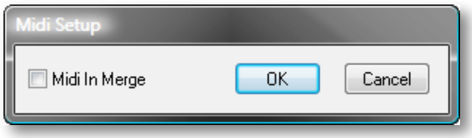
The entry **Automation Setup** opens the Automation Setup window.
All settings in that window have been described in detail
in Chapter 4 > Mixer Automation. Please refer to that section for details.

MIDI Setup



This section only relates to a system configuration that contains at least one legacy Soundscape 32 Unit.

The Midi Setup submenu only shows selectable entries for legacy Soundscape 32 hardware units .
Clicking on the Unit Entry opens a simple Dialogue:



If **MIDI In Merge** is activated (box checked), Soundscape V6 mixes the MIDI messages received at its MIDI input with the 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 this sequencer 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 (box unchecked).

Preferences

The section Preferences provides access to various settings affecting the general behaviour of the software and the way it interacts with hardware.
Four 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 V6 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 V6 detects audio material that is continuous across a cut (e.g., when a cut has been made in order to fade a section of a Part), and does not perform a deglitch in this case.

✓ Auto generate waveform files Ctrl+Shift+W

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.

✓ Keep all generated waveform files

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.

✓ Play till next Marker

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.

✓ Fast REW / FF

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.

✓ Scrub during REW / FF

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.

✓ Connect to Console Manager

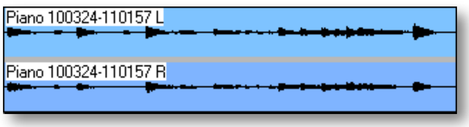
Activating the **Connect to Console Manager** option connects Soundscape to the Console Manager application, which it launches automatically if necessary. Deselecting it closes the Console Manager application.
Console Manager is a separate application and needs to be installed. **Chapter 2 › Installing Console Manager**

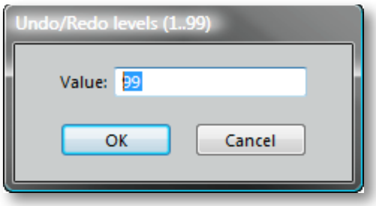
✓ Ignore incoming samplerate indication

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.

✓ Auto append DATE-TIME text to new Take names

When the **Auto append DATE-TIME text to new Take names** option is selected (ticked), recorded Takes get the current date and time appended to the end of their name in Year/Month/Day – Hour/Minute/Second format.

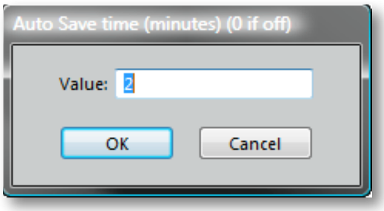




Clicking **Undo/Redo levels** opens a dialogue box where any value between 1 and 99 can be entered to determine the number of available Undo/Redo levels.

The Undo and Redo commands are found in the Edit menu.
The standard Windows [Ctrl]+[Z] and [Ctrl]+[Y] key 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.

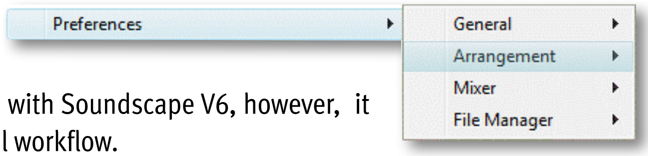
When multichannel audio is recorded as a 4 channel, 6 channel or 8 channel record block, imported or modified, an extension is appended to the Part name. The entry **Multi-Channel speaker name extension** can add a **numeric** extension to the name of each recorded file: 1_x, 2_x, 3_x..., where x is 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.

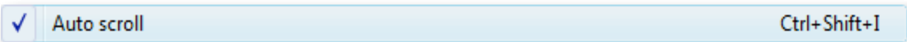
When the mouse pointer is on **AES/EBU (S/PDIF) Xmt Channel Status**, a submenu appears which lets you select the **Consumer** or **Professional** format for the digital audio data transmitted from a Soundscape 32's AES/EBU (S/PDIF) outputs.
This may be necessary due to the limitations of other devices. For example, some professional DAT recorders will not record from their digital inputs if they receive data in the "Consumer" format.

SSL Soundscape V6 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, when you are experienced with Soundscape V6, however, it is likely that you'll want to optimise certain settings for your personal workflow.

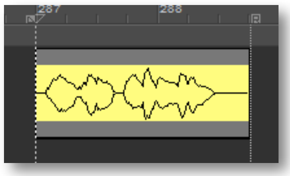


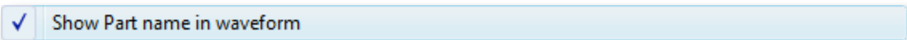


While by default the Current Locator moves in the Arrange window according to the current time position, if the **Auto scroll** option is selected, the Current Locator becomes fixed in the middle of the Arrange window, and the Arrangement itself (i.e., the Parts, Time Axis, and Marker Bar) scrolls in 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 the “Fill Waveform” parameter turned off (described later in this section), the waveform display will be redrawn faster when scrolling.
This behaviour is also known as "Tape Head" view in other applications.

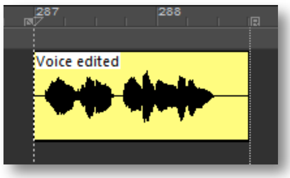
Selecting **Fast Redraw** mode will send the Waveform Display Graphics of each Track directly to the graphics accelerator card. Depending on the Speed of the PC and Graphics Card (especially the 2 D Performance of the Card) this can result in constant, partial Screen Updates, creating the impression 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 create a "smoother" screen redraw.

When the **Fill waveform** option is selected, waveforms are displayed as solid black. Otherwise they are displayed as outlines, which can speed up redraws.





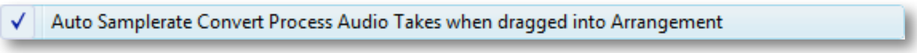
When the **Show Part name in waveform** option is selected, the name of each Part that displays a waveform or automation data is shown. Audio Parts that do not display a waveform do not respond to this setting, as their name is displayed anyway. Waveform based editing may be easier if the Part name is not shown in the waveform.



When the **Show Part volume** option is selected, the volume of each audio Part is represented by the vertical extent of the coloured section of the Part. If this option is disabled, the volume is still shown for any selected Part(s) or when certain tools (e.g., the Volume Trim tool) are used.

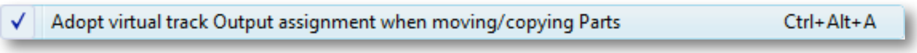
When the **Keep Part selection** option is active, Parts remain selected after they have been edited or processed. This is particularly useful when several editing and/or processing operations must be performed on a group of Parts, and re-selecting these Parts prior to each operation would involve zooming 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 in order to reflect changes to the Arrangement layout caused by a global "Cut & Insert" or "Cut & Delete" operation.

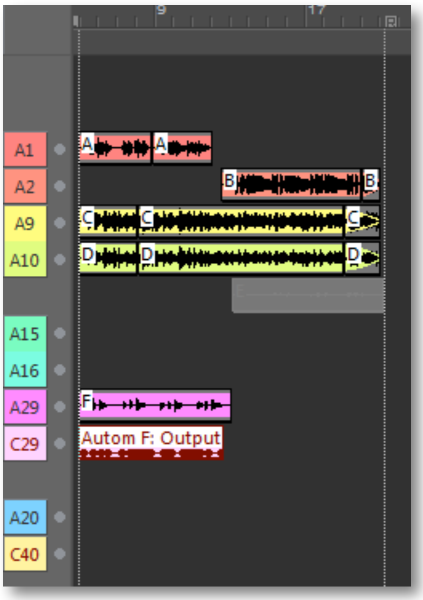


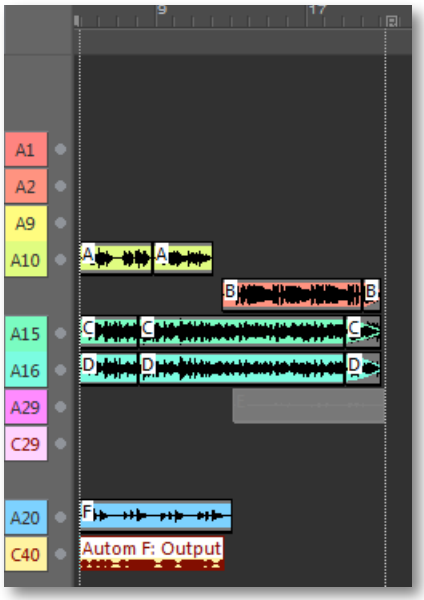
When **Auto Samplerate Convert Process Audio Takes when dragged into Arrangement** is enabled (ticked) and an audio Take is dragged from the File manager to the Arrange window, the Sample Rate is converted if necessary to match the Sample Rate of the Arrangement.

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).



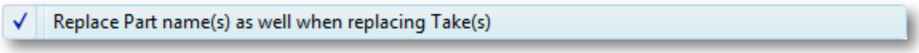
If **Adopt virtual track Output assignment when moving/copying Parts** is active, any record track inserted in a virtual track slot of the Record Track Column, also determines a default output assignment (physical track) for the corresponding virtual track.
Any Part moved or copied to a virtual track that has an output assignment is automatically assigned to the corresponding physical track, also Parts that are dragged from File Manager will adopt the Track assignment of the virtual Track they are dropped on..



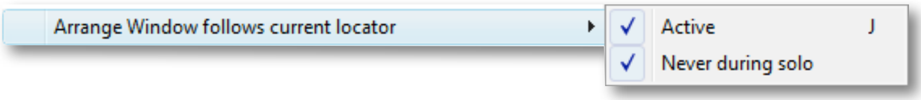


The group of Parts in the example on the left hand side is being moved down by three virtual tracks using the Move Vertical tool, with "Adopt virtual track..." active.

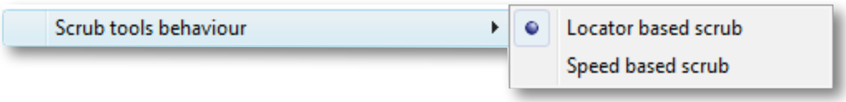
- Audio Part **A** has been assigned from track 3 to track 10.
  Audio Part **B** ends up on a virtual track that does not have an output assignment therefore it remains assigned to track 5.
- Audio Parts **C** and **D** (stereo linked, originally assigned to tracks 9-10) have been assigned to tracks 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** ends up on a virtual track assigned to output track 29. It remains muted but its track assignment is changed
- 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.



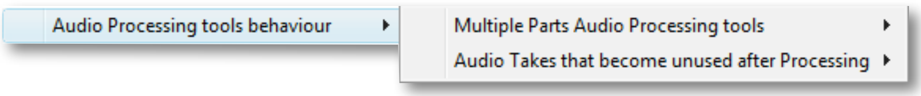
When **Replace Part name(s) as well when replacing Take(s)** is active, every time a Take referenced by a selected audio or automation Part in the Arrange window is replaced by right-clicking an appropriate Take in the File Manager and selecting "Replace Take(s)" in the context menu, the name of the 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.



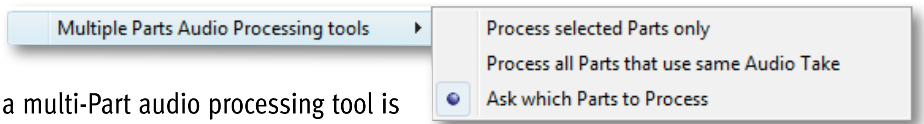
By default, the screen is redrawn to follow the Current Locator whenever it exits the visible section of the Arrangement.
When the mouse pointer is on **Arrange window follows current locator**, a submenu appears to activate or deactivate this function (which can also be toggled using the [J] key or 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 option **Scrub tools behaviour**, 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.**

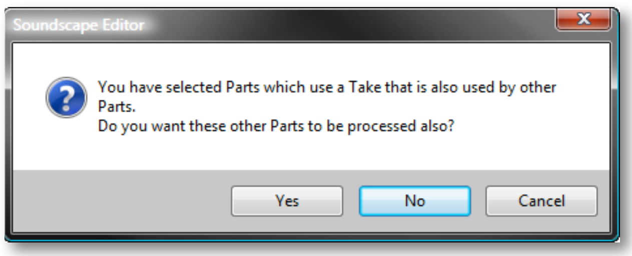


The option **Audio Processing tools behaviour** is divided into a submenu each for **Multiple Parts Audio Processing tools** and **Audio Takes that become unused after Processing.**



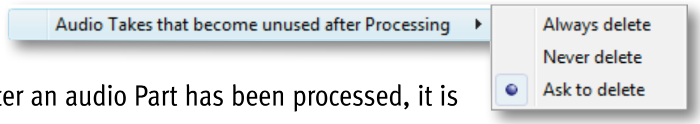
In the **Multiple Parts Audio Processing tools** submenu you have 3 options:

- **Process selected Parts only:** Whenever a multi-Part audio processing tool is used on a Part, 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:** Whenever a multi-Part audio processing tool is used on a Part, the processing is also applied to any other Parts that refer to the same audio Take.
- **Ask which Parts to Process:** Whenever a multi-Part audio processing tool is used on a Part, the following dialog box will be displayed, which lets you specify whether in this particular instance the processing should also be applied to other Parts that refer to the same audio Take.

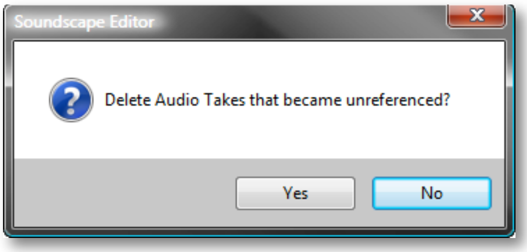


These settings only apply for the **Normalize** and **DC Removal tools**, which currently are the only Multi-Part Audio processing Tools.

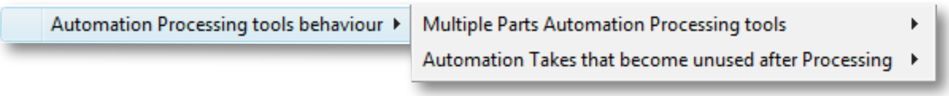
In the **Audio Takes that become unused after Processing** submenu you have again 3 options:



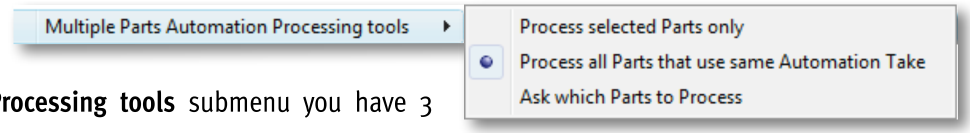
- **Always delete:** Whenever an audio Take becomes unused after an audio Part has been processed, it is deleted automatically without notice.
- **Never delete:** When an audio Take becomes unused after an audio Part has been processed, no action is taken. The Take remains on the disk.
- **Ask to delete:** Whenever an audio Take becomes unused after an audio Part has been processed, the following dialog box is displayed, which lets you specify whether in this particular instance the unused (unreferenced) Take should be deleted.



These settings apply for all audio processing tools.

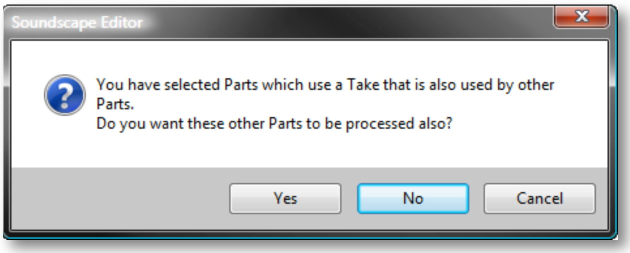


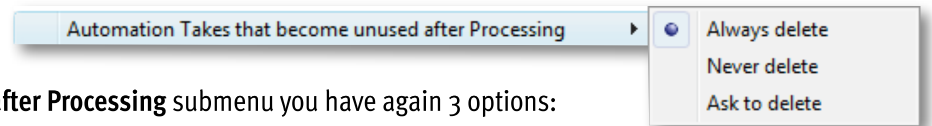
The option **Automation Processing tools behaviour** is divided into a submenu each for **Multiple Parts Automation Processing tools** and **Automation Takes that become unused after Processing.**



In the **Multiple Parts Automation Processing tools** submenu you have 3 options:

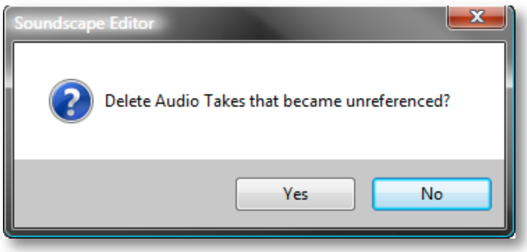
- **Process selected Parts only:** Whenever an automation processing tool is used on a Part, the processing is applied to that Part only. Other Parts that reference the same automation Take are not affected in any way.
- **Process all Parts that use same Automation Take:** Whenever an automation processing tool is used on a Part, the processing is also applied to any other Parts that refer to the same automation Take.
- **Ask which Parts to Process:** Whenever an automation processing tool is used on a Part, the following dialog box will be displayed, which lets you specify whether in this particular instance the processing should also be applied to other Parts that refer to the same audio Take.

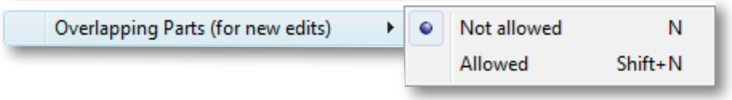




In the **Audio Takes that become unused after Processing** submenu you have again 3 options:

- **Always delete:** Whenever an automation Take becomes unused after an automation Part has been processed, it is deleted automatically without notice.
- **Never delete:** When an automation Take becomes unused after an automation Part has been processed, no action is taken. The Take remains on the disk.
- **Ask to delete:** Whenever an automation Take becomes unused after an automation Part has been processed, the following dialog box is displayed, which lets you specify whether in this particular instance the unused (unreferenced) Take should be deleted.





The option **Overlapping Parts (for new edits)** determines whether Parts in the Arrange window can overlap one another.
You can also type [N] or [Shift]+[N] to respectively **not allow** or **allow** Part overlaps.
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".

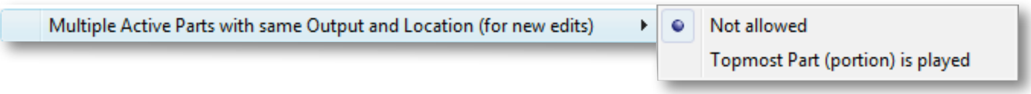
Overlaps are always allowed for crossfades (created using the Crossfade tool) regardless of the current setting.

If you select **Not allowed**, previously existing overlaps are preserved, hence the mention "for new edits". An overlap that has been suppressed (e.g., by moving a Part) can be restored by using the **Undo** function, even if the "Not allowed" setting 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 (e.g., 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 (e.g., 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 entry **Multiple Active Parts with same Output and Location (for new edits)** lets you determine whether multiple active Parts that overlap in time (regardless of their virtual track position) can have the same output assignment (i.e., 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.

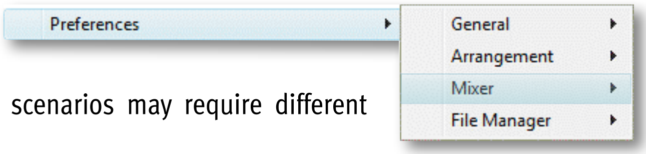
If **Topmost Part (portion) is played** is selected, any active Part can be assigned to any track without affecting the mute status of any other active Parts already assigned to that track. It is therefore possible to have overlapping active Parts with the same track assignment. If this is the case, 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.

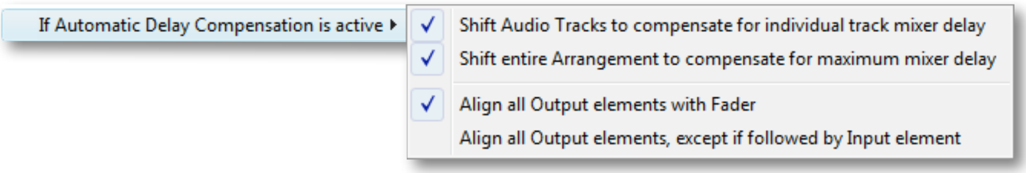
NOTES:

- 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.

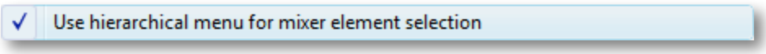


The Mixer Preferences include all options for the visual and logical behaviour of the Mixer Window.
The default settings are chosen carefully, however, certain work scenarios may require different settings.

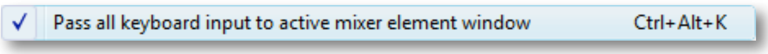
When **Auto route** is active, the DSP resources are allocated automatically to the various mixer elements, including plug-in effects, etc. When the option is inactive, these resources can be allocated manually by selecting the DSP cells for any Mixer Column upon creation. Mixer Columns can be manually assigned, even if this option is active, by clicking into an empty Mixer column area with the I/O Assign Tool.



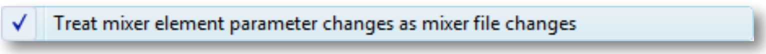
The heading **If Automatic Delay Compensation is active** leads to a further submenu, giving access to several optional settings relating to the operation of the Automatic Delay Compensation (ADC).
Please refer to **Chapter 4 › Mix Window › Automatic Delay Compensation (ADC)** for more details.



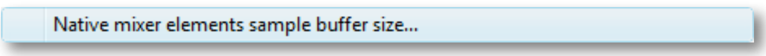
When the Create tool is used in Mixer Edit mode to insert a mixer element into a mixer column, a menu appears that lists all available mixer elements. The size of this menu can make it difficult to find a particular element, especially if a lot of plug-ins are installed or if a small monitor is used.
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 V6 may cease to work properly.



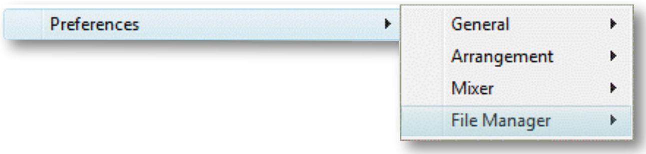
When **Treat mixer element parameter changes as mixer file changes** is enabled, a change to a mixer element parameter will enable the Save Mix function under the File menu. If it is disabled, only changes to the structure or routing of the Mixer enable the Save Mix function.

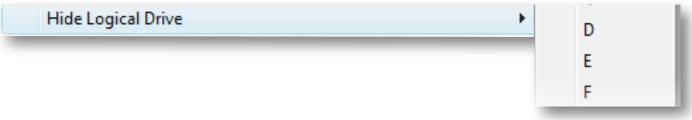


Clicking on **Native mixer elements sample buffer size** opens a Dialogue box where a value in samples can be entered for the native mixer elements buffer size. The default value is 128. Typically, it should be raised if the PC is struggling to cope with the processing demands.
Please refer to **Chapter 3 › Create your first Soundscape Project** for more details.

File Manager Preferences

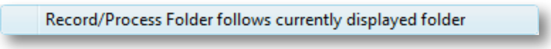
The File Manager Preferences includes 2 sub-menus:





The **Hide Logical Drive** option list allows you to hide logical 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 V6 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 activating **Record/Process Folder follows currently displayed folder,** Soundscape V6 is recording or processing 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.

This behaviour 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.

Various Settings

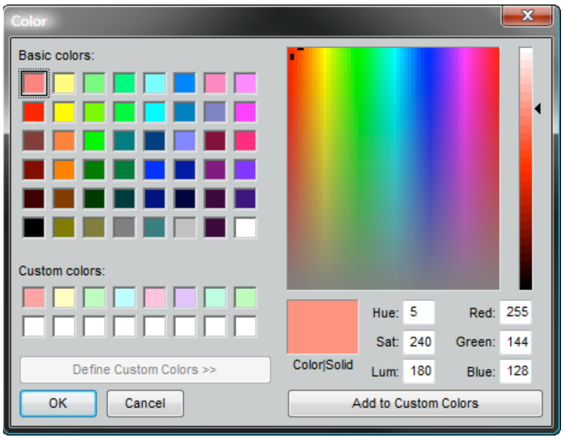
Muted Parts in the Arrangement and audio Takes in the File Manager can be played back or scrubbed through the Soundscape Mixer, with the Solo and Scrub Tools.

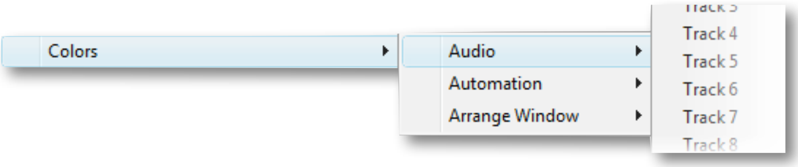
Clicking on **Muted Parts / Takes Solo Output › Audio**, submenus let you specify for each Unit in your system, which physical track should be used for playing back muted Parts and audio Takes.
This does not affect the current track assignments in the Arrangement, the chosen track is just “borrowed” as necessary 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.

Clicking on **Colors**, several submenus are displayed which can be used to access standard Windows colour palettes and customise the look of certain elements of Soundscape V6.

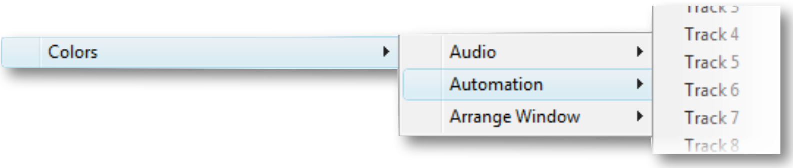
Since it can take quite some time to define new colours, ie. for 128 different Audio Tracks, remember to click **Save Settings** in the Settings Menu, otherwise your changes will be lost on next Program Startup!





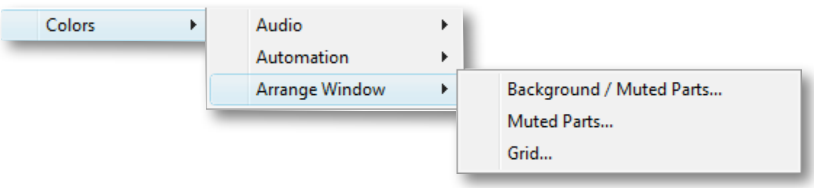
The colour setting of any audio Track can be accessed and edited by clicking the Track Number in the **Audio** sub menu.

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.



The colour setting of any automation Track can be accessed and edited by clicking the Track Number in the **Automation** sub menu.

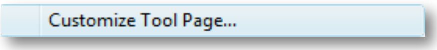
NOTE: Unlike the colours for Waveforms, the colour of the Automation Curves and Points adopt to changes of the automation Tracks background Colour. The underlying algorithm tries to create a best possible Contrast inside a similar colour range. For any background colour with a luminosity of around 128 (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, muted Parts only 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".

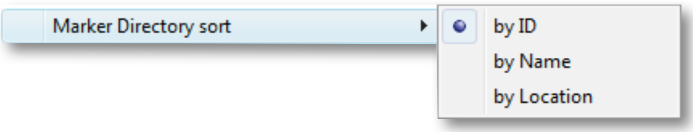
Customize Tool Page



Clicking on **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.

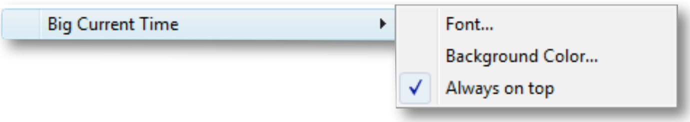
Marker Directory sort



With **Marker Directory sort** you can choose either 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.

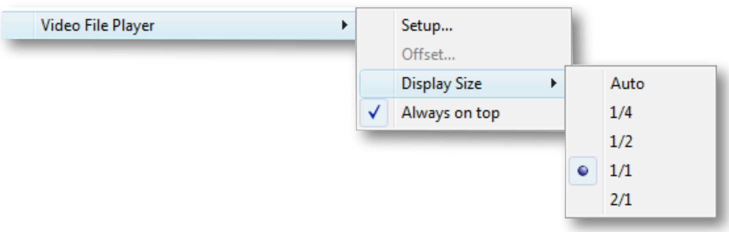
Big Current Time



The entry **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.

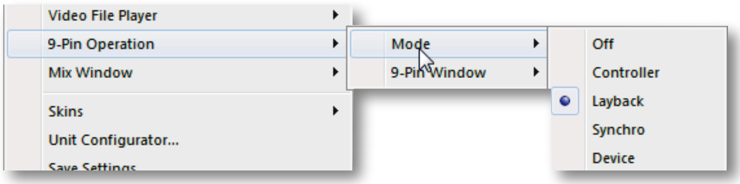
Video File Player



Video File Player opens a submenu which lets you access and define settings for Soundscape V6's integrated Video File Player.

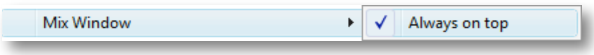
Please refer to **Chapter 4 > Video File Player Window** for a detailed explanation.

9-Pin Operation



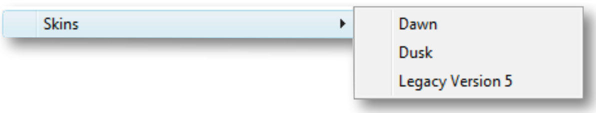
This item is only available if a Soundscape 32 with an installed Sync Option is part of the current System Configuration.
It appears greyed out if this is not the case.
The functionality and Setup of the 9-Pin Operation is described in Detail in **Chapter 4 > 9 Pin Window.**

Mix Window

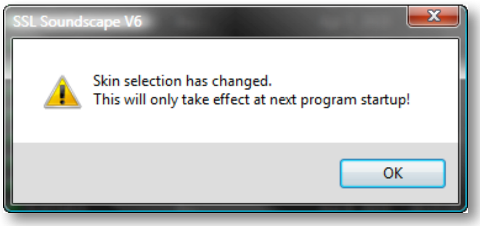


When the mouse pointer is on **Mix Window**, a submenu appears that lets you activate or deactivate the **Always on top** option. If selected, the Mixer window will always remain visible on top of any other window, even if that other window is active (except if that other window is also set to be "Always on top": in Soundscape, the 9-Pin Master, Big Current Time, and Video File Player windows can also be set this way).

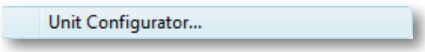
Skins



With the **Skins** option you may chose between different SSL Soundscape V6 looks. The skin selection changes most of the Arrange, File Manager and Main Window appearance, but keeps the Audio and Automation tracks colour settings intact.
After selecting a skin a message informs you that your selection will take effect at the next program startup:



Unit Configurator



Clicking this option opens the **Unit Configurator window** which can be used to change the Hardware System Configuration including all installed SSL MX4 or Mixpander units (under certain circumstances also legacy Soundscape Systems).

This utility is described in **Chapter 3 > Unit Configurator.**

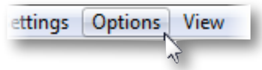
Save Settings

Clicking this item saves the current global Soundscape V6 settings available in the Settings menu.

Saving settings 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 made to the global Settings are temporary and will be reset to previous defaults when re-launching Soundscape V6, if Settings haven't been saved.

Clicking on Options opens the Passwords window, which lists all included or optional software plug-ins that are password protected and installed on the host computer.

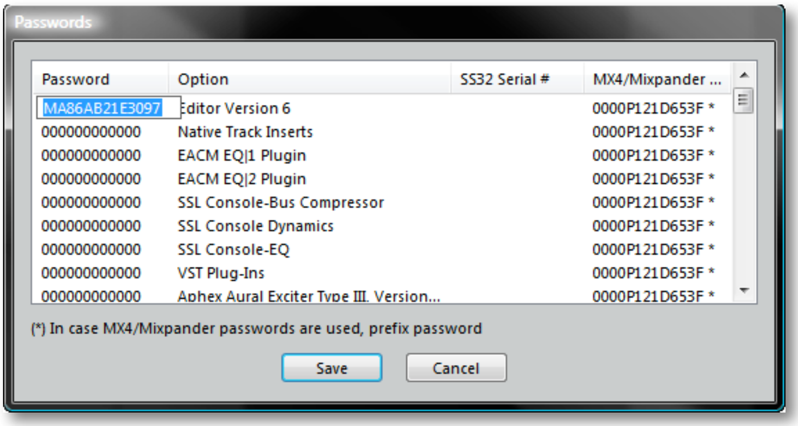


Please note that the SSL Soundscape V6 Software itself must be enabled by entering the correct password for the installed hardware unit.
Each SSL hardware unit has a unique serial number inside the DSP Hardware **(Unique ID or UID)**.

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.

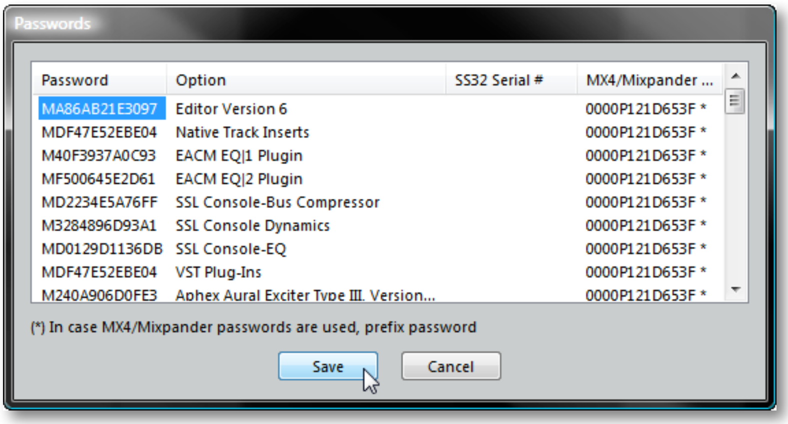
- The Mixer Password for a new SSL hardware unit can be found on a sticker that is placed directly on the board or on the product packaging and registration card. It can also be obtained from our website if you have registered the unit.
- For MX4, your Sticker should also contain Passwords for the 3 Plug-ins of the SSL Console Bundle and another Password for the the SSL Audio Toolbox.
- If you haven't purchased Soundscape V6, you can work with Soundscape V6 without any limitation in functionality (apart from the missing Time Module functions) and without any time limit, however Soundscape V6 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 SS V6;-).
- If you have purchased Soundscape V6 you have received the Soundscape V6 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.

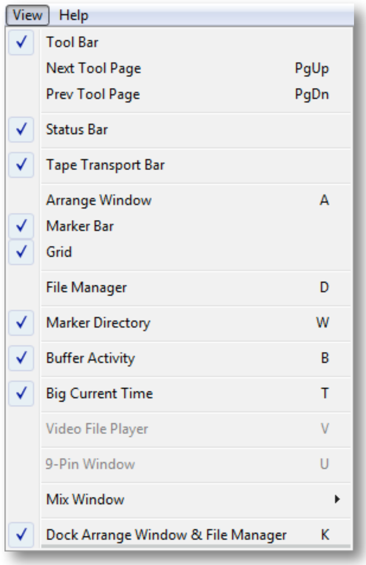
To enter a password simply double click on the password text field.



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 SSL Soundscape V6. 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, ProTools 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.

Soundscape V6 windows

- Main Soundscape window with Menu's, Toolbar, Status Bar and Transport Bar
- Arrange window with 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]

can be resized and positioned anywhere on Windows Desktop, including extended Desktops for Multi-Monitor Setups.

The Toolbar, Status Bar, Tape Transport Bar, and Marker Bar can be shown or hidden by clicking the corresponding menu entry.

- Buffer Activity window [B]
- 9-Pin window [U]

cannot be resized, but can be positioned freely.

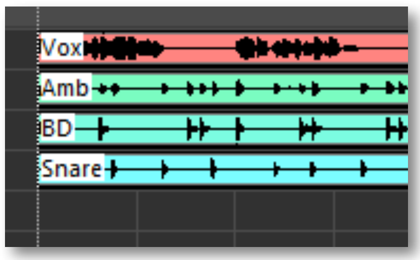
All screen settings including size and positions can be saved by clicking **Save Settings** in the Settings menu.

Next Tool Page/Prev Tool Page [PgUp]/[PgDown]

These two items can be used to scroll through the Tool Pages.

Grid

The entry will show or hide the Grid in the Arrange window.

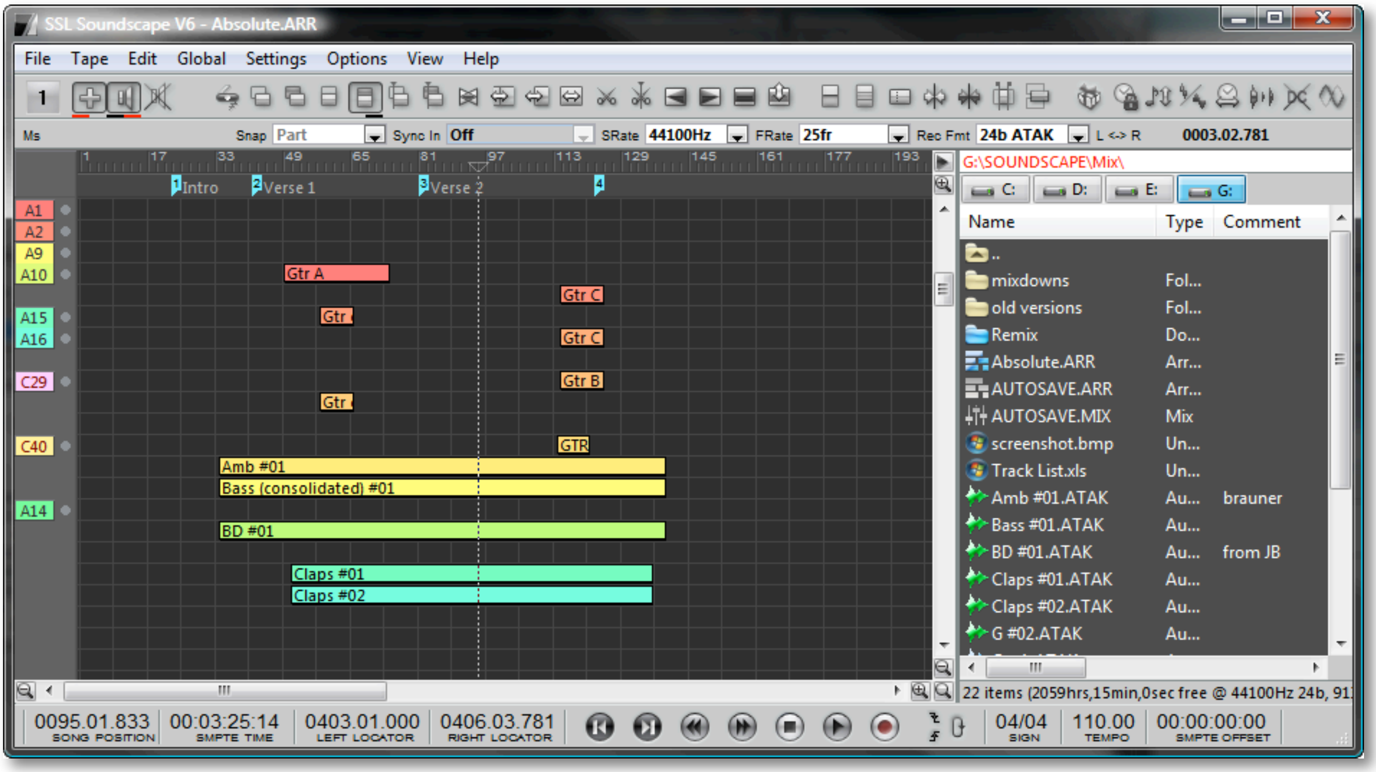


The Grid is composed of horizontal lines, which indicate virtual track positions.
The vertical lines indicate time subdivisions.

At the highest possible zoom level and if the Time Axis is set to SMPTE time in the Settings menu, one vertical line is displayed per frame.
If the Time Axis is set to bars/counts/ticks, one vertical line is displayed for every four ticks (there are 960 ticks per quarter note).

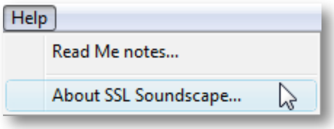
Dock Arrange window & File Manager

Selecting **Dock Arrange Window & File Manager,** docks both windows to Soundscape's Main window.
This function saves screen real estate and allows proportional adjustment of both windows while the main window is resized.
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 **Dock...**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, and the About SSL Soundscape Window, that displays additional System Information.

Also all Soundscape V6 related User Guides can be directly accessed from this Menu (including this User Guide).

User Guides
Clicking on any User Guide (Soundscape V6, Mixer V6, SSL Console Bundle, MX4 Hardware Installation) opens it in a PDF Format inside an appropriate reader application.

Please make sure that you have a PDF Reader installed on your System, otherwise Windows may display an Error message stating that it could not open this file Type.

Read Me notes
The Readme Notes contain the latest changes in software revision.
It may contain important last minute changes, that have not yet made it into the User Manual.

About SSL Soundscape
Opens a "usual" About Window, with Software Version Information, some licensing and copyright info and a nice and polished graphic:)

Interesting and often needed is 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, when you want to purchase additional software modules or want to register the hardware on our website.

Also 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.

2
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P 212
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File		Edit	
Open Arr	CTRL+O	Create	ALT+1
Append Arr	CTRL+A	Track Assign	CTRL+1
Save Arr	CTRL+S	Move	CTRL+M
Save Mix	CTRL+Q	Move Vert	CTRL+SHIFT+M
Exit	ALT+X	Move Loc	CTRL+ALT+M
Tape		Copy	CTRL+C
Go to Begin	Home	Copy Vert	CTRL+SHIFT+C
Go to End	End	Copy Loc	CTRL+ALT+C
Got to Prev Snap	SHIFT+LEFT	Cut	CTRL+X
Go to Next Snap	SHIFT+RIGHT	Cut Loc	CTRL+ALT+X
Go to Prev Part Border	CTRL+LEFT	Volume Tool	CTRL+V
Go to Next Part Border	CTRL+RIGHT	Vol Fade In	CTRL+E
Rew	LEFT	Vol Fade Out	CTRL+F
Fw	RIGHT	Vol Trim	CTRL+SHIFT+V
Stop	DOWN	Trim	CTRL+T
Play	UP	Trim Adjacent	CTRL+ALT+T
Rec	Num ADD	Scrub Solo	CTRL+B
Go to Left Loc	ALT+L	Scrub All	CTRL+SHIFT+B
Go to Right Loc	ALT+R	DC Offset	CTRL+SHIFT+N
Auto Punch In	CTRL+ALT+I	Normalize	CTRL+N
Auto Punch Out	CTRL+ALT+O	SRate	CTRL+SHIFT+S
Loop	Y	Pitch	CTRL+SHIFT+P
Arm Audio Trk	NUM MULTIPLY	Time Stretch	CTRL+SHIFT+T
DisArm Audio Trk	NUM DIVIDE	Reverse Tool	CTRL+SHIFT+R
		Mute	CTRL+U
		Glue	CTRL+G
		Crop	CTRL+3

Various Mouse + Key Actions	RClick=Right Click, LClick= LeftClick, DClick=Double Click
Create Record Track (Audio/Auto)	RClick 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	RClick+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
Move Current Loc to Marker	RClick 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	LClick on Time Axis Bar
Move Right Locator to here	RClick on Time Axis Bar
Move Current Loc to here	SHIFT+LClick on Time Axis Bar
Move Current Loc to L Loc	LClick on Left Loc Button (below Display)
Move Current Loc to R Loc	LClick on Right Loc Button (below Display)
Song Position (User Enter Value)	LClick on Song Position Time Display
Current Locator (User Enter Value)	LClick on SMPTE Time Display
Left Locator (User Enter Value)	LClick on Left Locator Time Display
Right Locator (User Enter Value)	LClick on Right Loc Button (below Display)
Sign (User Enter Value)	LClick on Sign Display
Tempo (User Enter Value)	LClick on Tempo Display

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P 212
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1 glue P 211 here

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Edit cont.		Global	
Noise Gate	CTRL+K	Cut/Ins L<->R	SHIFT+I
Waveform	CTRL+W	Cut/Del L<->R	SHIFT+D
Info	CTRL+I	Cut/Sel L<->R	SHIFT+C
Delete	CTRL+D / DEL	Cut at Current Loc	SHIFT+F
Stereo Link	CTRL+L	Move selct‘d Parts to L Loc	SHIFT+Q
Multi-Ch Link	CTRL+4	Copy selct‘d Parts to L Loc	SHIFT+W
Slip	CTRL+J	Select all Parts	SHIFT+A
Invert Phase	CTRL+P	Select all P‘s same virtual TRK	AHIFT+V
Repeat	CTRL+R	Select all P‘s same OUT	SHIFT+O
Timelock	CTRL+SHIFT+L	Select all P‘s same TAKE	SHIFT+T
Auto-Curve Select	CTRL+SHIFT+A	Select all P‘s Part to End	SHIFT+G
Auto Event Edit	CTRL+SHIFT+E	Select all muted Parts	SHIFT+M
Thin Auto Events	CTRL+SHIFT+F	Select Timelocked Parts	SHIFT+K
Context Sensitive	CTRL+2	Select P‘s L<->R Loc	SHIFT+S
Snap Point Edit	CTRL+SHIFT+D	Select P‘s L Loc -> End	SHIFT+R
Mixe Edit	E	Unselect all Parts	SHIFT+U
Auto Eneble	G	Invert Part selection	SHIFT+X
ADC	CTRL+ALT+D	Set L+R Loc start/end sel P‘s	SHIFT+L
Auto Snapshot	F	Set L Loc sel P‘s end	SHIFTE
Zoom IN H	ALT+RIGHT	Set R Loc sel P‘s begin	SHIFT+B
Zoom OUT H	ALT+LEFT	Flatten Arrangement	CTRL+SHIFT+K
Zoom IN V	ALT+UP	Compute Tempo L<->R	CTRL+SHIFT+J
Zoom OUT V	ALT+DOWN	**Settings**	
Scroll L	CTRL+ALT+LEFT	Sync In = OFF (Master)	M
Scroll R	CTRL+ALT+RIGHT	Sync In = MTC (SS32)	S
Scroll U	CTRL+ALT+UP	Sync In = LTC (SS32+Sync Opt.)	O
Scroll D	CTRL+ALT+DOWN		

Selct Multiple Parts in Arr	Hold Down Ctrl, LeftClick on Parts
Selct all Parts from Sel Prt to End	Hold Down Shift & Control, LeftClick 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
(... to enable an Inactive Mixer)	DClick Mix Window caption bar
Select tool to L or R mouse button	LClick or RClick on tool in ToolBar
Select tool to Alt+L or Alt+R mouse	ALT+LClick or ALT+RClick on tool in ToolBar
Up a tool page	RClick on ToolBar page icon
Down a tool page	LClick on ToolBar page icon
Customize Context Sensitive Edit Tool	DClick on SwissArmyKnife Icon
Customize Cross-Fade Settings	DClick on CrossFade Tool
Customize ToolBar Settings	DClick on blank part of ToolBar
Zoom Box (for Arr... to zoom in)	LClick Zoom Tool-> draw zoom
Zoom Out (for Arr... to zoom out)	DClick ZoomTool
Magnify Waveform display in Part	RClick on Zoom IN Vert
Un-Magnify Waveform display in Part	RClick on Zoom OUT Vert
Select Multiple SOLO buttons in Mixer	Hold CTRL while LClick or RClick solo buttons
Select default response to Pop Up‘s	Spacebar or Enter
Cancel Processing Tools	Spacebar or Enter

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glue P 213 here 3

2 glue P 212 here

Settings cont.	
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+G**
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**
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**

Nudging with Move or Copy on left Mouse	
1 Track Up	**LClick+UP**
1 Track Down	**LClick+DOWN**
Previous Snap Point	**LClick+LEFT**
Next Snap Point	**LClick+RIGHT**
Trim Tool on left Mouse	
Trim Part Previous Snap Point	**LClick+UP**
Trim Part to Next Snap Point	**LClick+DOWN**
Slip Tool on left Mouse	
Slip Part Previous Snap Point	**LClick+LEFT**
Slip Part Next Snap Point	**LClick+RIGHT**
Snap Point Edit Tool on left Mouse	
Reset Snap Point	**DoubleClick on Part**
Volume&FadeTool / 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	**RClick on Part**
ZOOM	**RClick Part, move mouse**
Scroll Part in Arr Window	**ALT+RClick, move mouse**

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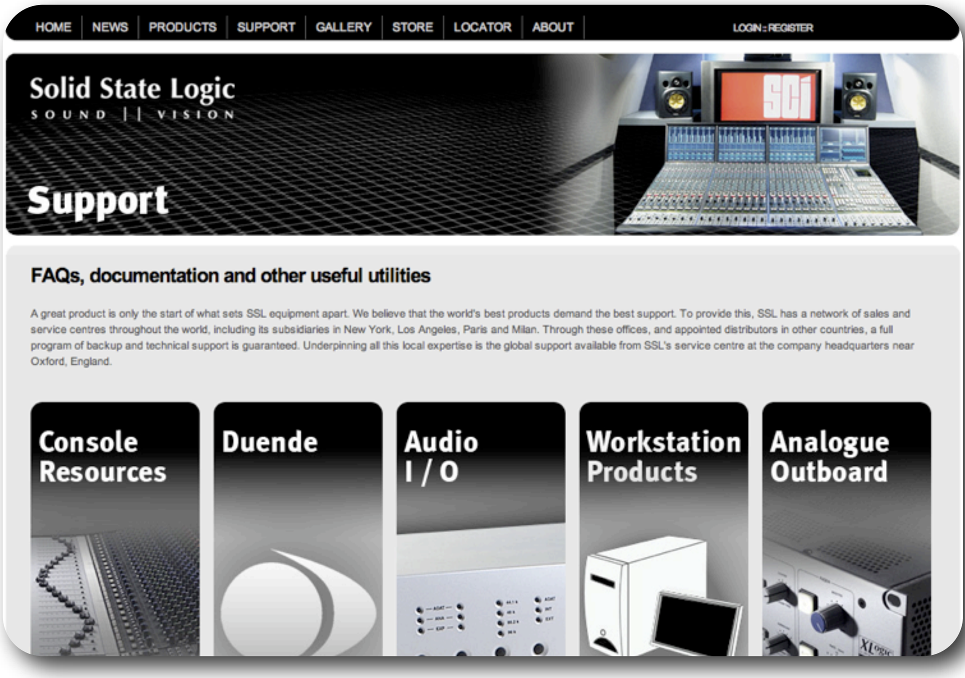
Support, FAQs and Online Help Centre

To access the latest support information on SSL Soundscape and Mixer V6, 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.
All information 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.

URL: **http://www.solidstatelogic.com/support**



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Visit SSL at: **www.solid-state-logic.com**