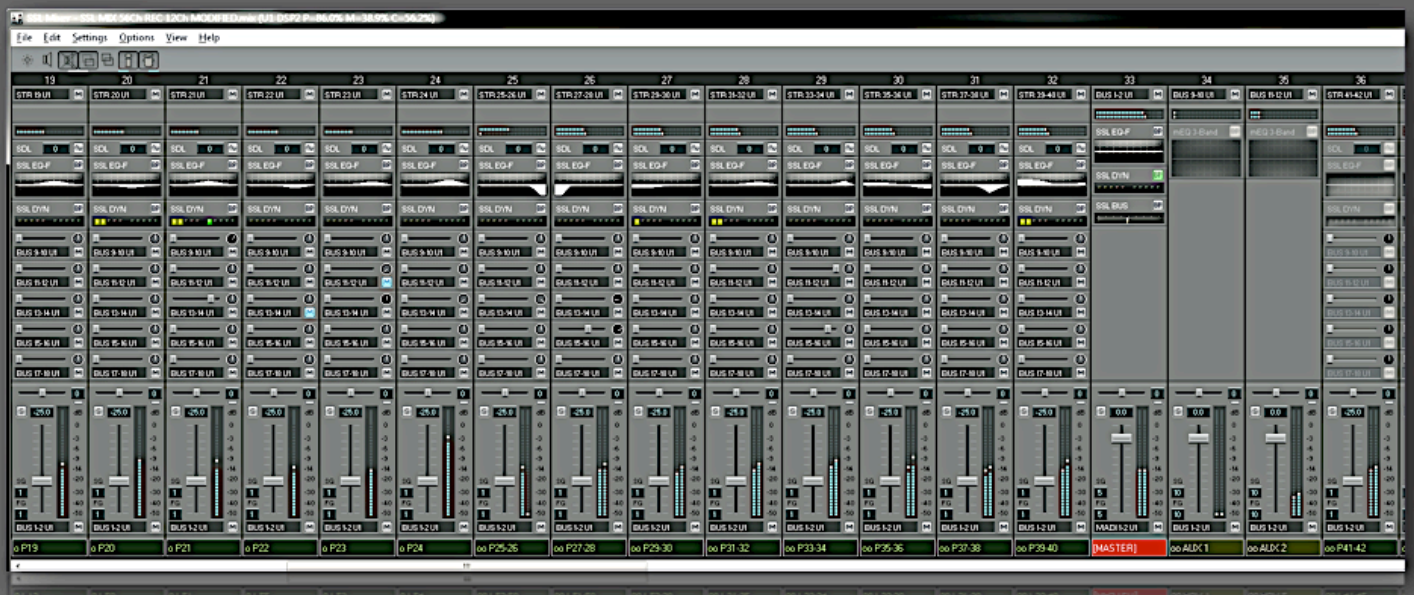
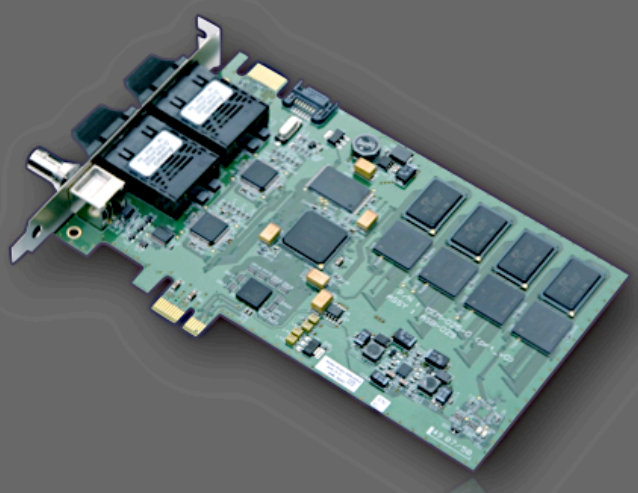
www.solid-state-logic.com



SSL Mixer V6

SSL Console Magic **in the box**. For Windows PC's.

Manual and Reference Guide V6.01



SSL Mixer V6. This is SSL.

Solid State Logic
SOUND || VISION

Table of Contents

1. Introduction

Congratulations on your purchase of an SSL MX4 audio card.

The SSL MX4 is designed to provide you with many years of reliable service and, in conjunction with the SSL XLogic Alpha-Link converter series, 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 on our website. This will ensure that you receive notifications of future software upgrades and other important information, and that your guarantee is registered. Registration will also make you eligible for technical support. The Solid State Logic home page is at: **www.solid-state-logic.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 Mixer V6 is the Software creating a "real" SSL Console inside the Box.
Flexible, reliable and equipped with the Sound that made more Hit Records than any other Console.

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, the SSL Mixer 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, the Mixer is fully configurable, and while a few ready-made Mixers are provided, to really harness its power, read the **"Mixer" chapter** 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 Mixer software's routing possibilities. 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 mixer configurations. 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.

Please also read the MX4 Hardware Manual for detailed Information about Hardware Installation and Driver Setup.

If possible: Please do not print this manual.

NOTE: 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 Mixer 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.

Key commands and key combinations
Some functions of the SSL Mixer 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 SSL Mixer 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 the SSL Mixer can vary (e.g., the Toolbar can be moved around the main window). The screenshots themselves were not always made using the SSL Mixer's default settings.

Safety and Installation Considerations

This section contains definitions, warnings, and practical information to ensure a safe working environment. Please take time to read this section before installing or using this unit. Please do not dispose of these instructions.

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

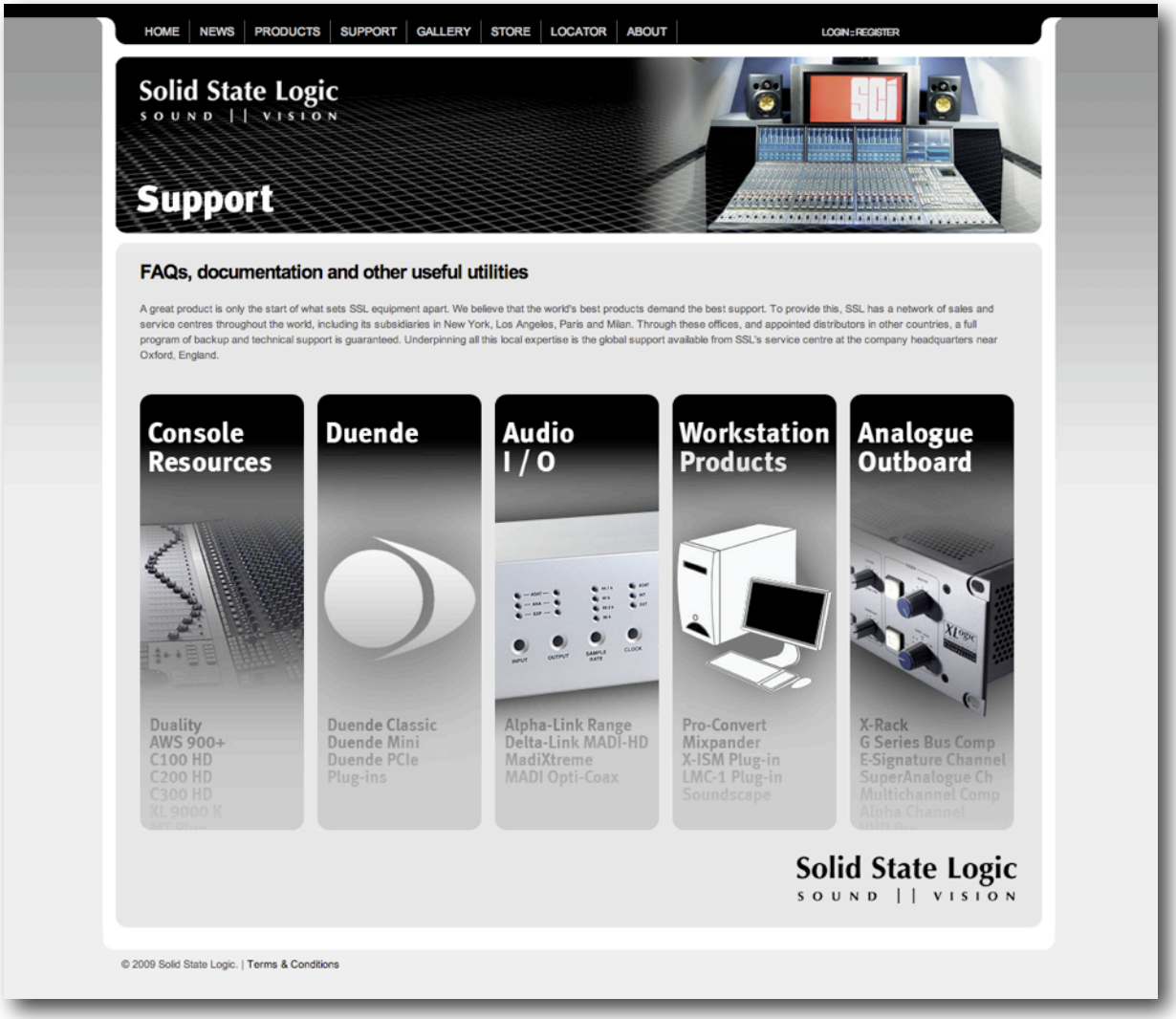
This manual has been written with great care and attention to detail, and we have attempted to cover every operational aspect of the SSL MX4. However, it is not a contractual document. Solid State Logic and/or the writer(s) of this manual cannot be held responsible for any loss or damage arising directly 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.solid-state-logic.com**
The **SSL Support Website** is: **http://solidstatelogic.com/support**



2. SSL Mixer V6 System Requirements

Compatible SSL Audio Cards

MX4

MX4 PCIe card features two MADI inputs and two MADI outputs, providing 112 or 128 simultaneous inputs and outputs at up to 48kHz, 56 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 SSL Alphalink interfaces or SSL Soundscape iBox range of audio interfaces.

Compatible Windows Versions and Driver Protocols

The **SSL MX4 PCIe** card and associated 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	

Note: MX4 can run under OSX Tiger V10.4.11, OSX Leopard V10.5.4 or greater and OSX Snow Leopard V10.6.1 (32 and 64 Bit) or greater 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 Mixpander PCI** card and associated software are compatible with the following operating systems and driver protocols:

Platform	Operating System	Driver Protocols
PC	Windows XP SP2 or later (32bit only)	ASIO 2,WDM,MME,GSIF2,DWave
	Windows Vista SP1 or later (32bit only)	

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.

MADI Converter, Console or Routing Device

As world leader in large format analogue and digital mixing consoles, it was natural for SSL to be among the companies that initiated the development of MADI (Multichannel Audio Digital Interface).
The MADI standard was finalised by the AES as AES10, first published in 1991 and revised in 1993. A PDF copy of the specification can be obtained from the AES website.
With outstanding digital audio workstation systems and innovative DSP-powered multichannel audio cards, the SSL product line has always been at the cutting edge of professional studio technology in terms of sonic performance, processing power and multiple inputs/outputs. Featuring SSL's revolutionary PCIe-Core Audio Pipeline technology and supported by the legendary SSL Soundscape drivers, MX4 builds on this heritage as one of the first audio devices to take full advantage of the high speeds and incredible bandwidth offered by the PCIe bus on modern computers.

3. The 'SSL Inside' Console: SSL Mixer V6

Digital mixing, effects and processing

The SSL Mixer 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 in the way the channels are structured.

The SSL Console EQ-Filters, Channel Dynamics, and Bus Compressor plug-ins offer the highest quality processing you can find in digital audio and provide console grade processing for the 'hit record'.
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 in the signal path, so much so that 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 a "plug-in delay compensation" feature. This solves the problem in mixing situations, but is, by its very nature, unusable while recording, tracking and monitoring.

In contrast, the SSL Mixer and its 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, into your chosen application.

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. The SSL Mixer supports the VST format. This means that VST effect plug-ins running on the host CPU can be inserted directly in the SSL Mixer. This is useful in situations where latency is not an issue (e.g., during mixing and mastering) and allows access to a multitude of plug-ins.

Feel free to grow: Multiple card setups

Multiple MX4 and Mixpander cards can be used simultaneously to provide a staggering number of inputs, outputs, buses and audio streams with an enormous amount of processing power. All the units are controlled from the same instance of the SSL Mixer software and the different cards can be combined freely. In short, whatever your needs, you can choose a SSL configuration to match them.

Studio friendly

The SSL hardware is friendly to the rest of your studio too. You can hook up your external effects units to the SSL Mixer using any number of pre/post sends and insert additional mixer strips to mix the return signal. Every element in the mixer is in fact a plug-in that you can place exactly where required, which means that you can record to disk through the EQs, compressors or other effects.

If you already have a mixing console or you use the native EQs from your sequencer or audio software, then use the power of the SSL hardware for the cool DSP effects rather than straightforward EQs and compressors. It's all up to you, and it means that you do not end up paying for processing power that you do not need, as is the case with some PCI cards that offer an inflexible mixer, with no possibility of running plug-in DSP effects from third party developers. All settings and the entire mixer configuration can be saved and loaded from a PC file, so you never need to lose that perfect mix you were working on yesterday!

4. SSL Mixer V6 Software Installation

SSL Mixer V6 Software and Plug-In Installation Overview

Before you proceed installing any **SSL Mixer V6 or associated application software**, please ensure that you have successfully finished the Hardware Installation, as described in **the MX4 Hardware Manual.**
Otherwise you will not be able to successfully Install or work with the SSL Mixer V6 application.

The files required for the following Installation can either be found on the MX4 CD 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. Install the SSL Mixer V6 Software.
3. Install the Plug-Ins: SSL Console Bundle Plug-Ins **(This is part of No.2 if you use the MX4 CD Mixer Installer!).**
4. If you want to remote control the SSL Mixer V6 with a Hardware/Fader Remote, please also install the Console Manager Plug-In.
5. Start the SSL Mixer V6 Software and follow the Instructions of the Unit Configurator. Make sure the Software runs properly.
6. Enter the Mix Password you find on your MX4 Registration card, under Options|Passwords|MIX starting with the letter M.
7. Enter the SSL Console EQ Password you find on your MX4 Registration card (SEQ), under Options|Passwords|SSL EQ-Filters starting with the letter M.
8. 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.
9. Enter the SSL Bus Compressor Password you find on your MX4 Registration card (SBC), under Options|Passwords|SSL Bus Compressor starting with the letter M.
10. 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.
11. Connect your Converter or MADI Console and set Clocking and MADI Mode under Settings|Master Clock and Settings|MADI| MADI Mode.
12. Ensure that the Mixer 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.

Installing the SSL Mixer V6 Software

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**.

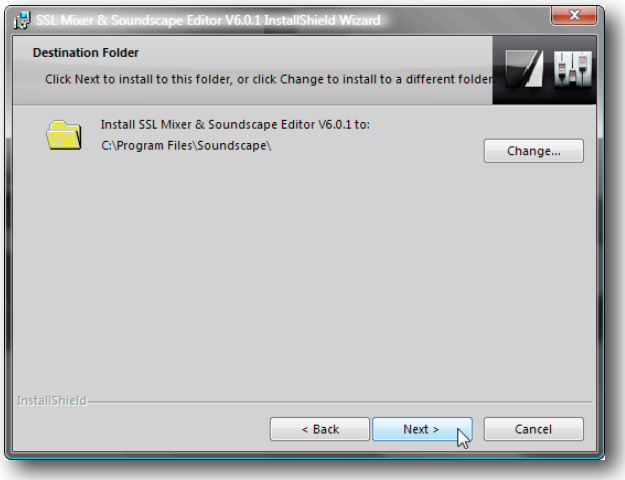
If your MX4 Installation CD has a Version prior to June 2010, or you are looking for a more recent Software Version, please **download the latest** Mixer/Soundscape V6 Combo Installer from our Website: **http://www.solidstatelogic.com/music/soundscape/downloads.asp**
Simply unpack the download into a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

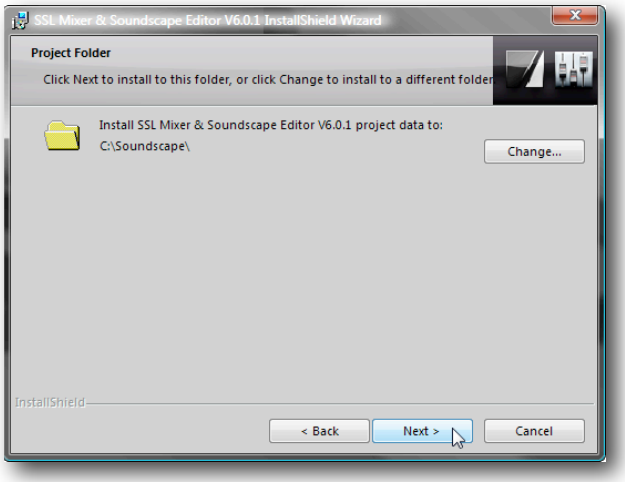
Please note that although you can select a different installation folder than the default offered, the SSL Mixer & Soundscape V6 user manuals will always refer to the default installation folders.
Also our Support docs and Support people will always refer to the default directories...

We therefore recommend using the default installation locations.



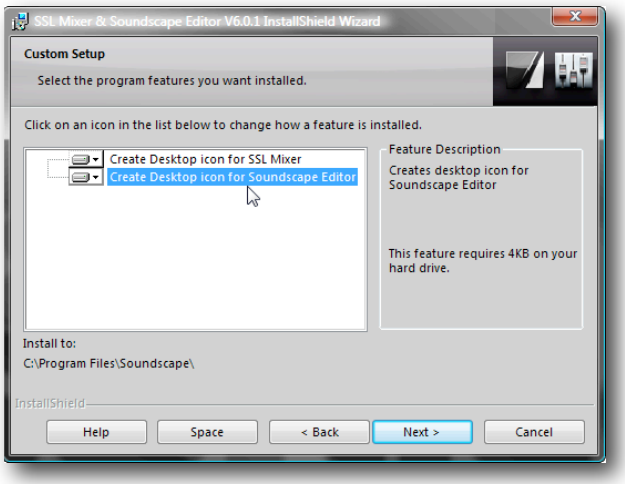
Click **Next.**

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



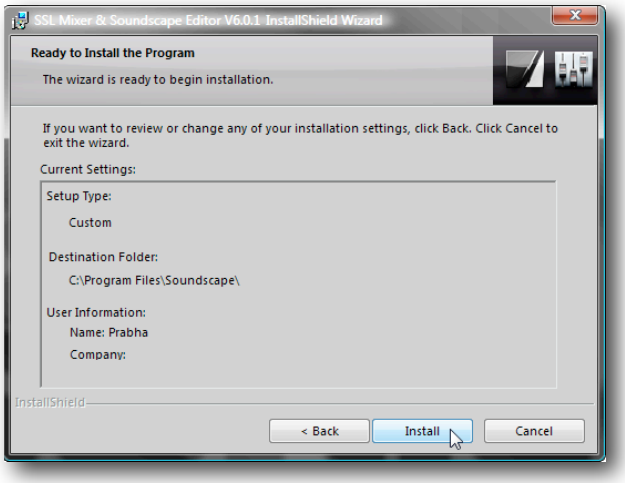
Click **Next.**

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 the SSL Mixer V6 for the first time...

Enjoy....

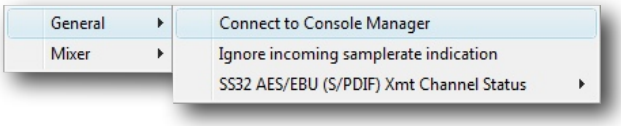
Note: The SSL Mixer /Soundscape V6 Combo Installer will also install the latest Version of the **SSL Console Bundle** and **Audio Toolbox**.

Installing the SSL Soundscape Console Manager Module

Double Click the **Setup Console Manager 1.55** to install the Console Manager software on your computer. Please follow the on screen instructions of the Installation wizard.

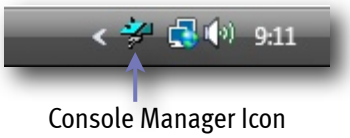
Setup Console Manager 1.55

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.



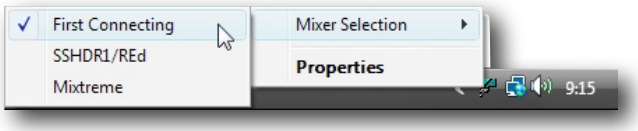
NOTE: If you always want to invoke the Console Manager Module when you start up the SSL Mixer V6, please activate **Connect to Console Manager** followed by a **Save Settings** command in the Menu **Settings|Save Settings**.

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



Console Manager Icon

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 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.

5. Quickstart Tutorial

Introduction

Unlike "simple Soundcards" the MX4 doesn't only provide sophisticated Audio I/O for your DAW Application to communicate to the outside world in large amounts, but also puts the coolest invention since sliced bread right into the heart of your Studio: An SSL Console "in the box".

In order to use this "Console" as your main or secondary DAW Mixer, the MX4 needs to become the 'communicator' between your DAW Application and the SSL Mixer V6 Software.

Before we start to glue everything together, please make sure you have:

1. Installed at least one MX4 SSL audio card in your PC and connected any required interface or external device (Alpha-Link, iBox or a MADI mixing console).
2. Installed the SSL Device Drivers.
3. Installed the SSL Mixer software.
4. Connected a stereo sound source to the SSL hardware's audio inputs 1/2
5. Connected the SSL hardware's audio outputs 1/2 to a monitoring system.
6. You are now ready to use the SSL Mixer V6 together with your DAW application.

Please follow each step carefully, and if you need further assistance check the Table of Contents. **Here we go...**

1. Boot your PC
2. Launch the SSL Mixer V6
3. Click Options menu to open the Passwords window. Enter your password for the "Version 2.xx Mixer" (MIX), Audiotoolbox (ATB), SSL Console EQ and Filters (SEQ), SSL Console Dynamics (SCD) and SSL Bus Compressor (SBC).
   The Passwords for your MX4 can be found on a label on the card itself or on the registration card.
   If you have purchased any optional effects plug-ins, then you can also enter the corresponding passwords now. Please check the "Passwords" section of the "Options" chapter if you need more help.
4. Click "Open" under the File menu, locate and load the Mixer "SSL_Console_16CH.mix" file.
   The path is C:\Soundscape\Mix\ SSL_Console_16CH.mix for a standard installation using the default folders.
5. 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. Make sure the red "clipping indicators" are not lit, otherwise you could get some very nasty digital distortion.

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 for details.

6. The Mixer will be set up to mix the 16 inputs to stereo and send the resulting stereo signal to outputs 1/2 of your SSL hardware.
7. Try loading different mixer files (menu: File|Open) to get an idea of how versatile the SSL Mixer is. Note that some of the preset mixer files will not load correctly if: they are created for a different Card or for a different sample rate, particular plug-ins are not installed yet or if the corresponding plug-in passwords have not been entered.
8. You are ready to start using the MME, ASIO-2 or GSIF drivers with your application(s). Alternatively, you can move on to the **Chapter 8. SSL Mixer V6 Reference Guide**, which describes how you can create or edit your own mixer configurations.

Working With your Audio Application

In order to use your MX4 most effectively, you will need to be familiar with the operation of your DAW or other audio capable application(s).
If necessary, please spend time reading the manual(s) for your application(s) of choice. In particular, if you are using a sequencer, make sure you understand the basics such as setting up the clock synchronisation, configuring the inputs and outputs of the software mixer, arming tracks etc.

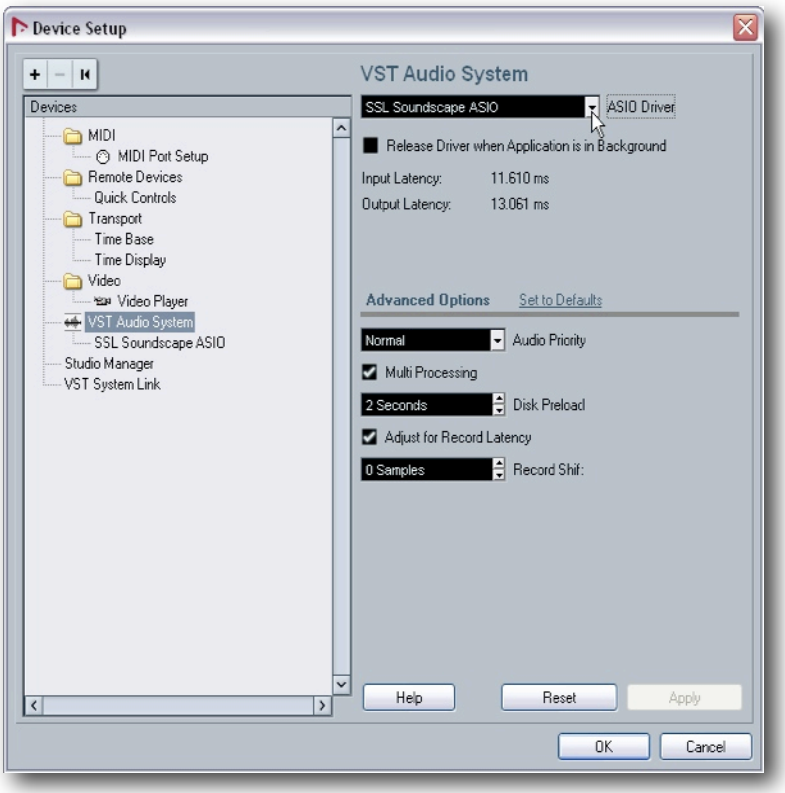
This manual could not possibly cover every aspect of every audio application on the market.
In the following sections you find a guide for some popular applications. The information should be similar for many other audio applications.

Achievable latency may vary depending on the host computer system and the applications that are running. During testing, we have found that MX4 operates very well at buffer settings down to 64 samples with many audio applications.
Please refer to the documentation of your audio application if you need information about selecting a buffer latency setting.

Steinberg Cubase/Nuendo

(Screenshots based on Nuendo 4.1.2 Build 851 running under Windows XP)

Selecting the Driver
For this quick-start example, we have chosen to use the ASIO driver (Devices menu, click Device Setup, highlight VST Audio System in the left-hand pane, select **SSL Soundscape ASIO** in the ASIO Driver drop-down menu then click OK).



Audio Connections and Clock Settings
We will assume here that the MX4 is connected to an Alpha-Link MADI AX or MADI SX.

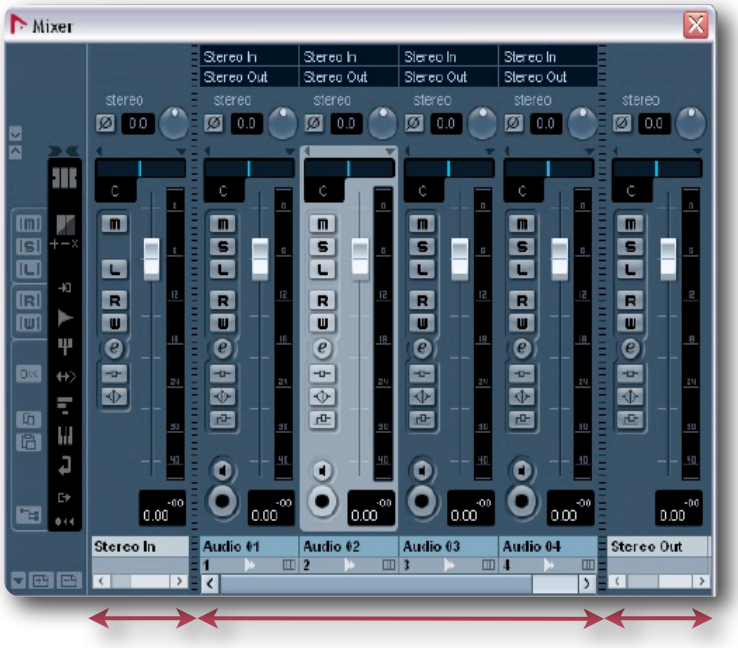
The Alpha-Link is configured so that its analogue inputs 1 to 24 are connected to its MADI outputs 1 to 24, and vice versa.

A stereo test signal is being received via the Alpha-Link's analogue inputs 1 and 2, and its analogue outputs 1 and 2 are connected to a monitoring system.

It is further assumed that the MX4 and Alpha-Link clock settings are correct.

The Nuendo Mixer is divided into three main sections (press **function key F3** to display the Mixer):

- **Input channel strips** (or 'input busses'), to the left of the mixer
- **Output channel strips** (or 'output busses'), to the right of the mixer
- **Audio channel strips** (in the centre section of the mixer)

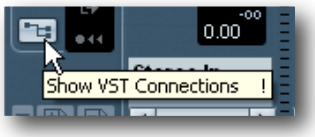


Input Section **Centre Section** **Output Section**

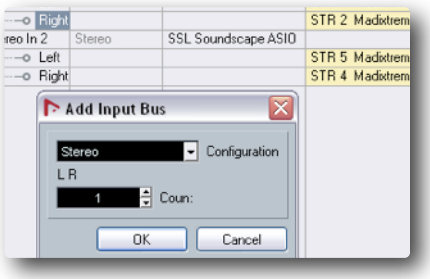
- Input and output channel strips are created automatically when an input or output bus is created in the VST Connections window.
- They are assigned automatically and incrementally to the MX4's inputs and outputs, but can be reassigned manually if required.
- Audio channel strips are created automatically in the centre section of the mixer when audio tracks are created in the current Nuendo Project.
- These audio channel strips can receive signal from any input bus channel strip or from the audio track they were created for, and transmit it to any output channel strip.

Input Channel Strips

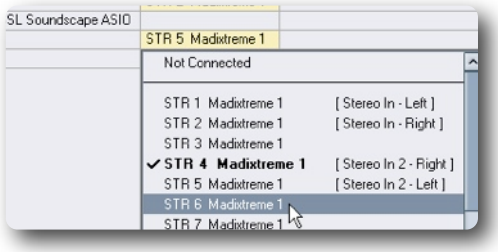
- Click the **VST connections button** (bottom left of the mixer) to open the VST Connections window.



- In the VST connections window, check that the Inputs tab is selected.
- Click the **Add Bus** button and use the resulting dialogue box to create input channel strips as required.

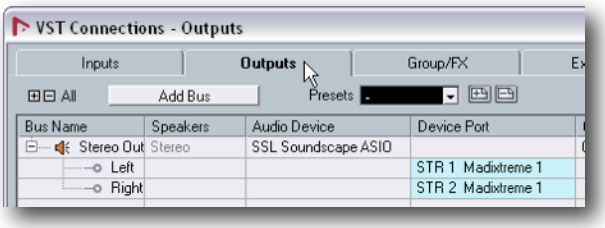


- To assign an existing Nuendo input bus to a different MX4 input, click the name of the MX4 input in the Device Port column and select the required input from the resulting menu.



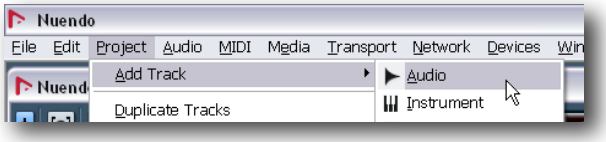
Output Channel Strips

Output channel strips are created and can be reassigned in the same way as input channel strips. Simply click the Output tab to add or assign output busses as required.



Audio Channel Strips

- In the Project menu select Add Track and click Audio.



- In the following **Add Audio Track dialogue box** create a number of audio tracks as required. The corresponding channel strips will be added to the mixer.

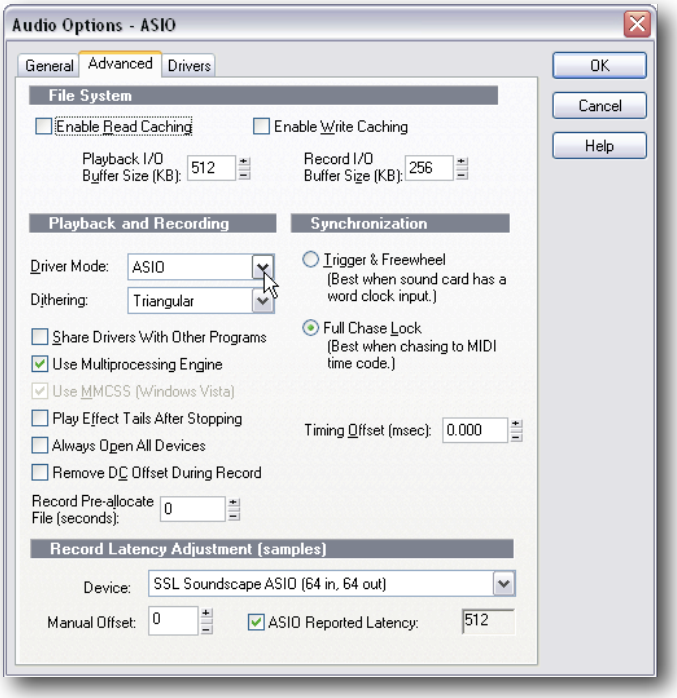
The input and output channel strips that are currently connected to the audio channel strips are indicated in **routing boxes located above the audio channel strips**.
Click the boxes to connect the audio channel strip to a different input or output channel strip as required.

If the **boxes are hidden,** right-click the mixer, select Window and then click **Show Routing View** to display them.

(Based on Sonar 7.0 Producer Edition running under Windows XP)

Selecting the Driver

The ASIO driver mode was chosen for this quick-start example (Options menu, click Audio, click the Advanced tab, select ASIO using the Driver Mode drop-down menu and click OK to confirm and close the window).



Audio Connections and Clock Settings

It is assumed that MX4 is connected to an Alpha-Link MADI AX or MADI SX.
The Alpha-Link is configured so that its analogue inputs 1 to 24 are connected to its MADI outputs 1 to 24, and vice versa.

A stereo test signal is received via the Alpha-Link's analogue inputs 1 and 2, and its analogue outputs 1 and 2 are connected to a monitoring system.

It is further assumed that the MX4 and Alpha-Link clock settings are correct.

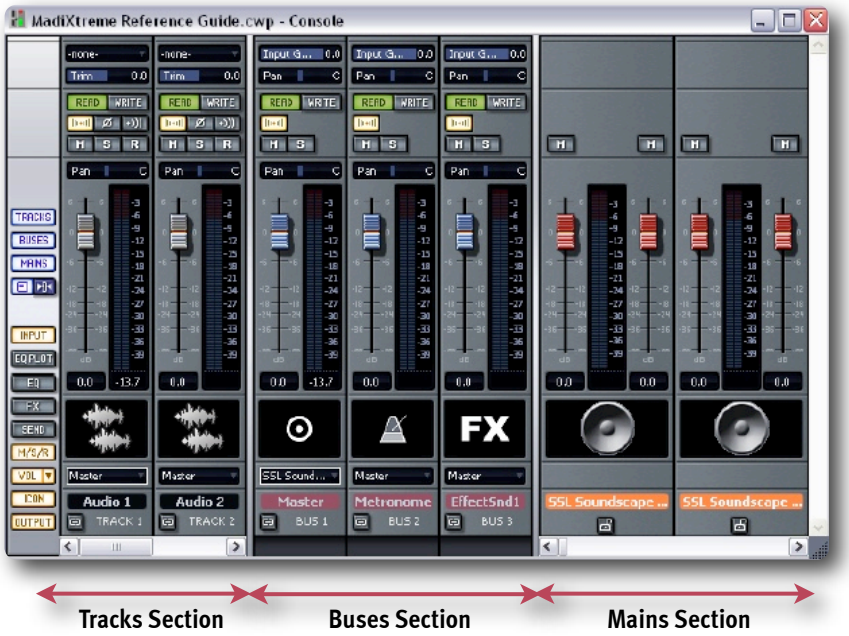
Overview

The Sonar Console is divided into three main sections (on the keyboard, press Alt + 3 to display the Console).

For clarity, we have used the buttons on the left of the console to hide the EQ Plot, EQ, FX and Send sections of the channel strips.

- Tracks Section: includes one channel strip for each track in the project
- Buses Section: includes the Master Bus strip on the left
- Mains Section: includes direct stereo outputs assigned to the audio hardware

When Sonar is started, a dialogue offers the option to create a new project. The default configuration of the new project is two stereo audio tracks and two MIDI tracks. This is a good way to start experimenting with MX4 and was used for writing this manual, so please follow the prompts to create a new project and then proceed to the sections below.



Selecting an Input for the Track/Channel Strip

In order to hear the input signal an input must be selected for the channel strip. Click the input field near the top of the track channel strip to display a selection menu.

If you are receiving a stereo test signal via MX4 inputs 1 and 2, highlight SSL Soundscape ASIO. A submenu will be displayed: click Stereo SSL Soundscape ASIO STR 1 MX4 1 in the submenu.

Selecting an Output for the Track/Channel Strip

Next, select the track output by clicking the output field near the bottom of the track channel strip.

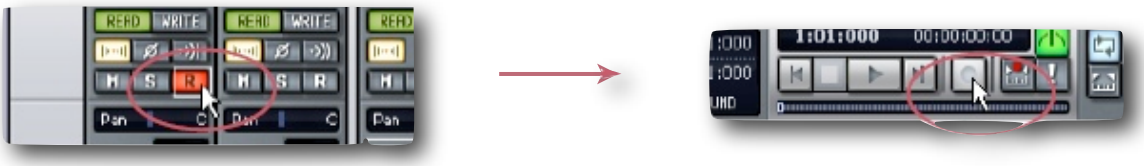
Select SSL Soundscape ASIO STR 1 MX4 1 to hear the signal transmitted via the MX4's first output pair.

Monitoring the Signal

Click the Input Echo button to hear the signal going through the channel strip prior to recording

Recording

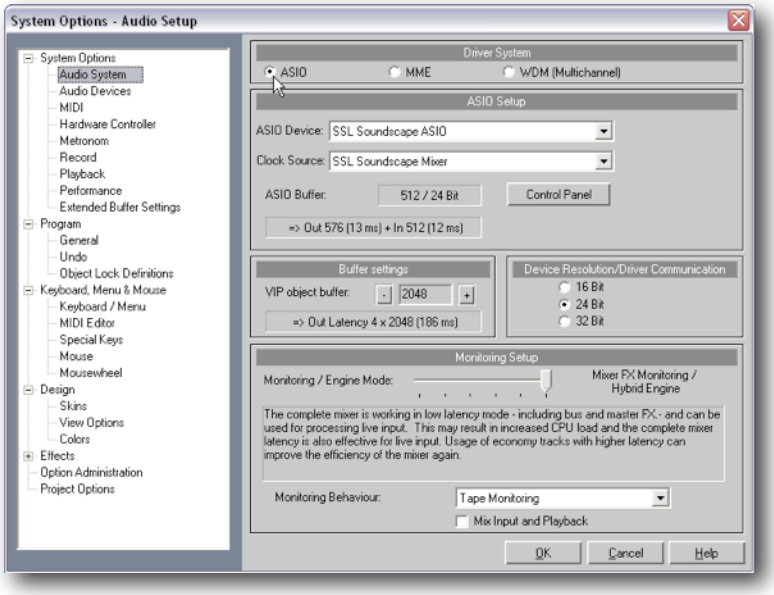
When the signal is passing through the channel, you can record it by clicking the Track Arming button in the track channel strip, then the Record button in the Transport:



(Based on Samplitude 10 Pro running under Windows XP)

Selecting the Driver

The ASIO Driver System was selected for this quick-start example (Options menu, click System/Audio, select the ASIO option box. Click **OK** to confirm your selection and close the window.



Audio Connections and Clock Settings

It is assumed that the MX4 is connected to an Alpha-Link MADI AX or MADI SX.
The Alpha-Link is configured so that its analogue inputs 1 to 24 are connected to its MADI outputs 1 to 24, and vice versa.
A stereo test signal is received via the Alpha-Link's analogue inputs 1 and 2, and its analogue outputs 1 and 2 are connected to a monitoring system.
It is further assumed that the MX4 and Alpha-Link clock settings are correct.

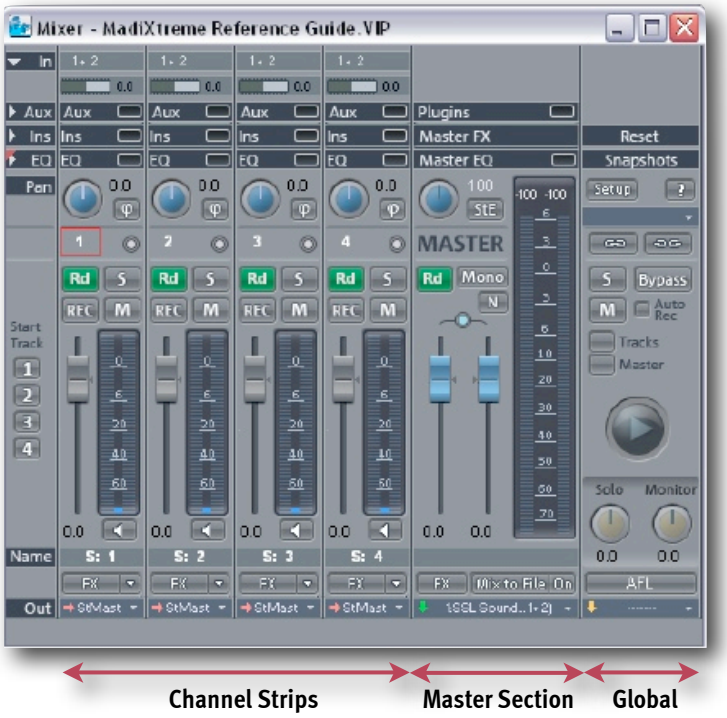
Overview

The Samplitude Mixer is divided into three main sections (on the keyboard, press M to display the Mixer).

For clarity, we have used the buttons on the left of the Mixer to hide the EQ Insert and Aux sections of the channel strips.

The Samplitude Mixer features one audio channel strip for each audio track in the current project.
By default, the output of each audio channel strip is routed to the Master section – this can be changed if required.
To get started, create a new Samplitude project by clicking New Virtual Project (VIP) under the File menu. Name the project in the following dialogue window and click **OK** to confirm and close the window.

By default, the project will have four stereo audio tracks and corresponding audio channel strips. We want to record our test stereo signal to track 1.



Selecting an Input for the Track

In order to hear an input signal the track input must be selected.

Click the input field near the top of the track channel strip to display a selection menu.

If you are receiving a stereo test signal via MX4 inputs 1 and 2, make sure that SSL Soundscape ASIO In (1+2) is selected (as it should be by default).

Selecting an Output for the Track

Next, select the track output by clicking the output field near the bottom of the Track channel strip. Make sure that

1. SSL Soundscape ASIO Out (1+2) (Stereo Master) is selected (as it should be by default).

Note that the output signal is not routed directly to the MX4 outputs 1 and 2. It is routed to the Master section of the Samplitude Mixer, which itself should be routed to the MX4 outputs 1 and 2.

If you need to change the output routing of the Master section, click the output field at the bottom of the Master section, just like you did for the output of the channel strip, and select the output pair as required.

Monitoring the Signal

Click the Input Monitoring button to hear the signal going through the channel strip prior to recording. The track channel strip meters will indicate the signal level, and you will hear it through your monitoring system. If necessary, adjust the level of your audio source.

Recording

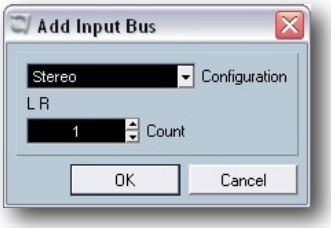
When the signal is passing through the channel, you can record it by clicking the Track Arming button in the track channel strip, then the Record button in the Transport:

Example configuration for Mixing, Recording and Monitoring with SSL Mixer V6 and Nuendo

In order to fully understand the benefits of DSP based Mixing, please read the following example carefully.
It was done using Steinbergs Nuendo DAW Software. When you are using a different Software and are already familiar with it, it is very easy to "translate" the Nuendo specifics to the procedures the DAW of your choice may require.

Preparing Nuendo

For Multitrack Mixing and Recording through the SSL Mixer V6 you will need to create enough Output and Input buses in Nuendo repeating the procedure described inside the Nuendo Section above.



You can add multiple buses at the same time in the **Add Input Bus/Add Output Bus Dialogues.** You can find these dialogues in Nuendo's **Device|VST Connections Menu,** using the information provided below (I/O Bus set up).
Selecting the MX4 as Nuendo's ASIO Audio device connects all I/O buses to MX4 Streams.

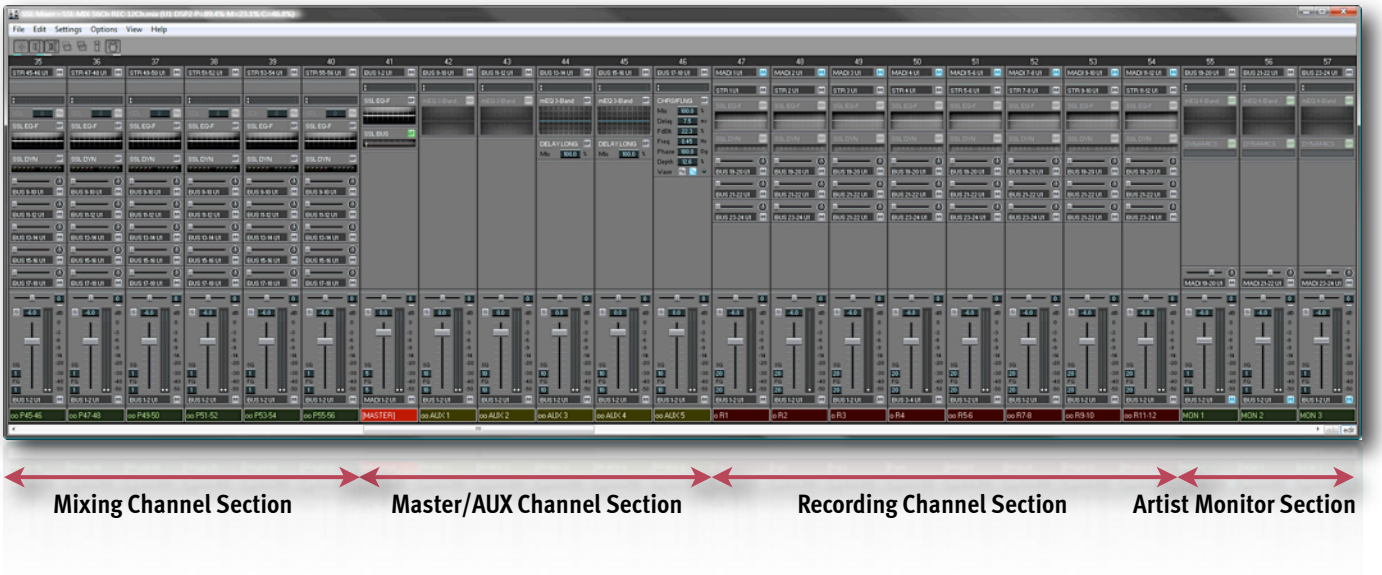
I/O Bus set up

Please create the following Inputs and Outputs in Nuendo:

- **6 Mono input buses** (and corresponding input channel strip) assigned to STR 1 to STR 6.
- **3 Stereo input buses** (and corresponding input channel strip) assigned to STR 7-8 to STR 11-12.
- **24 Mono output buses** (and corresponding output channel strip) assigned to STR 1 to STR24.
- **16 Stereo output buses** (and corresponding output channel strip) assigned to STR 25-26 to STR55-56.

Preparing SSL Mixer V6 | Loading the example Mixer

Switch to SSL Mixer V6 (ie. with ALT+Tab) and **Open** the Tutorial Mixer **SSL MIX 56Ch REC 12Ch .mix** in ../Program Files/ Soundscape/SSL Mix V6/Mix/Mixer/**SSL MIX 56Ch REC 12Ch .mix**



This Mixer is divided into a **Mixing Channel, a Master and AUX Channel, a Recording Channel and an Artist Monitor Section**.
It is therefore well suited for a Mixing Session with additional Overdubs.

On the following pages the Channel structure inside the different sections is explained in more detail.
Please read the descriptions carefully to fully understand the work- and signal flow.
It might be a good idea to read and work with the Mixer at the same time.

56 Inputs (24x Mono and 16x Stereo) are set to receive 56 Streaming Inputs from Nuendo's 56 Outputs.

Right below the input there's a **blank space** left to insert another **DSP or VST plug-in**.

Followed by a **Sample Delay Line** which can can invert the phase and add a small delay (0-255 samples) to change phase relations between channels.

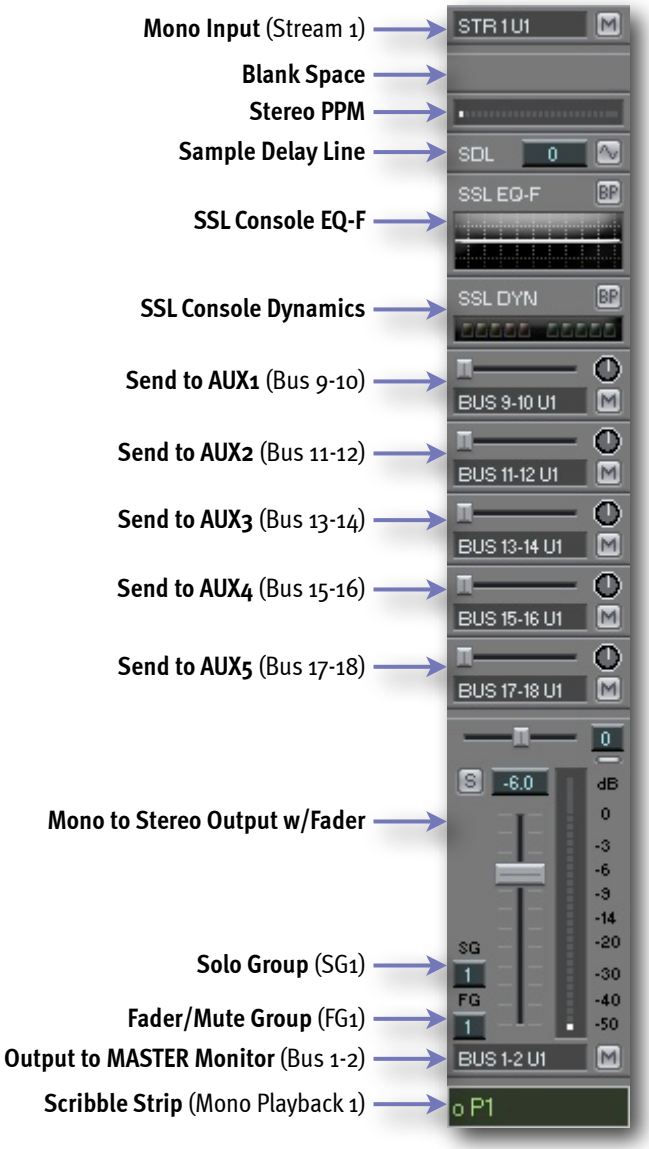
SSL Console EQ-F and Dynamics give you the original SSL Console Sonic DNA.

5 Stereo Sends can feed the AUX channels.
All AUX sends are set to **pre-fader** but may be switched to **post-fader** by double clicking them.
Sends can be placed before or after EQ and Dynamics with the **MOVE Tool** (Edit Mode Tools) if desired.

Solo and Fader Groups (SG and FG) are assigned to Group 1.

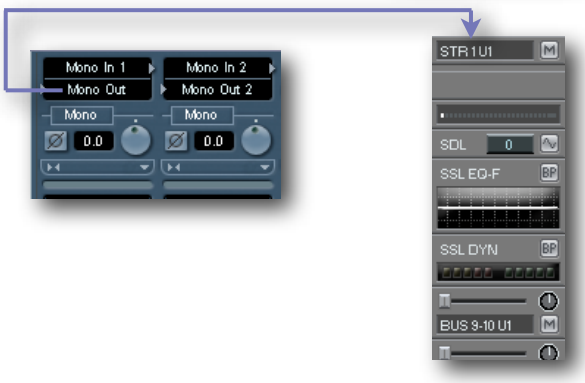
The **Output is set to Bus 1-2,** which is the Input of the MASTER Monitor.

Please note: Some Plug-Ins in the higher numbered Stereo Mixing Channels are **DSP muted,** to save DSP resources.
Although **MX4 has immense processing power,** it is a good practice to DSP mute unused elements to keep this DSP resource in reserve..



Interaction with the DAW Application

Every Output Bus in Nuendo now connects directly to the streaming inputs of the Mixing Channels.



The Mixing Channels have been set up in a very similar way to a traditional mixing console, Input, Input Metering, Delay Line/Phase Switch, EQ and Filters, Channel Dynamics and AUX Sends.

Since the SSL Mixer has no fixed structure you may change the Strip to completely meet your current demands.
You may add as many elements as you want and change its order in any way you can imagine, place additional AUX Sends or move the Dynamics above/before the EQ, simply create your own ideal Mixing Channel.

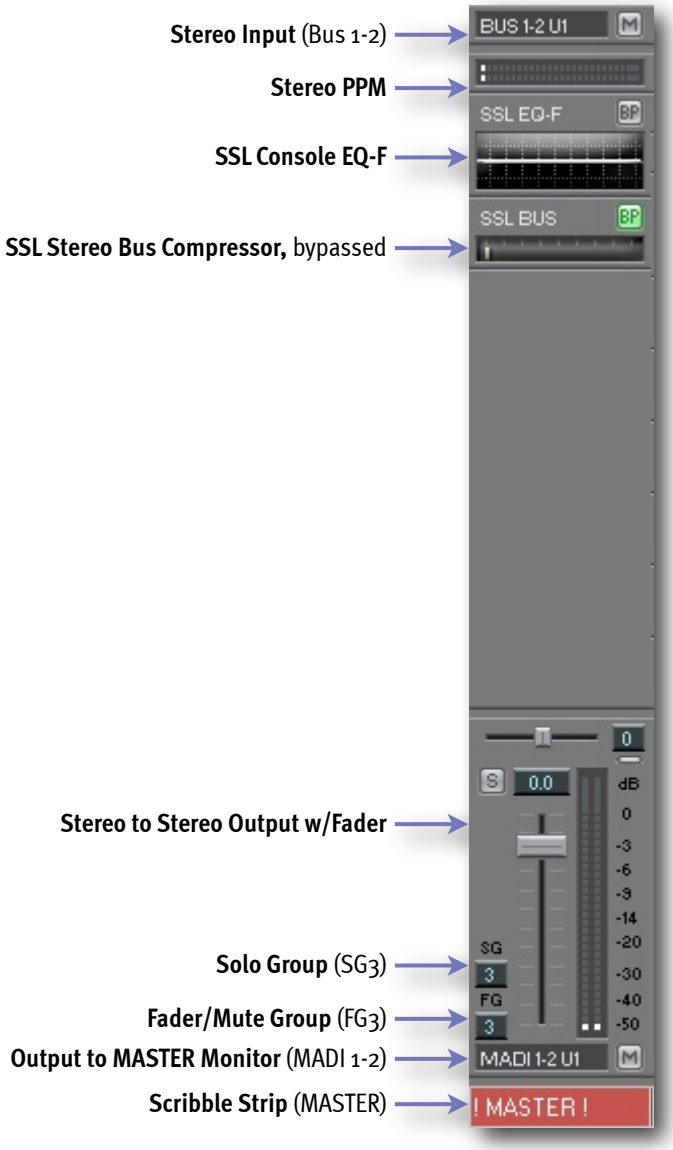
In the Tutorial mixer **SSL Mix 56Ch Rec 12Ch,** the flow diagram between the Mixing Channels, the Master Strip and the Aux Channels is quite simple.
Please proceed to the next page, where the **MASTER Channel** is explained in more detail.

The Stereo Master is used as the Main Monitor output. This should be routed and connected to your Monitor speakers, via the DA converter outputs.

The Input is set to BUS 1-2, the Outputs of all other Channels are set to Bus 1-2, so this Channel works as the summing section for all other columns in this Mixer.

The Stereo PPM meter shows any overload on the BUS.
Since the SSL Realtime Mixer Engine does not allow or mask any potential overloads in Inputs or Outputs, you may have to lower the general volume of all Mixing Channels proportionally by right clicking and dragging one of the faders to move them all.
Just below the PPM there is a Stereo SSL EQ-F for final Frequency tweaks followed by "The Glue" , our famous SSL Bus Compressor to let your Mix "sound like a record".
You could add other Plug-Ins to your Master Channel, for example a brick wall Limiter, a Multi-Band Compressor or a Mastering Package. Also you can add Sends or Outputs for example "To Tape" (Streaming Output to Nuendo to record the Master there) or to your Headphone Preamp, directly under the Plug-Ins.
The master channel's Solo and Fader groups are set to group 3 so it won't be affected by the solo buttons or group fader movements in the Recording or Playback channel sections.

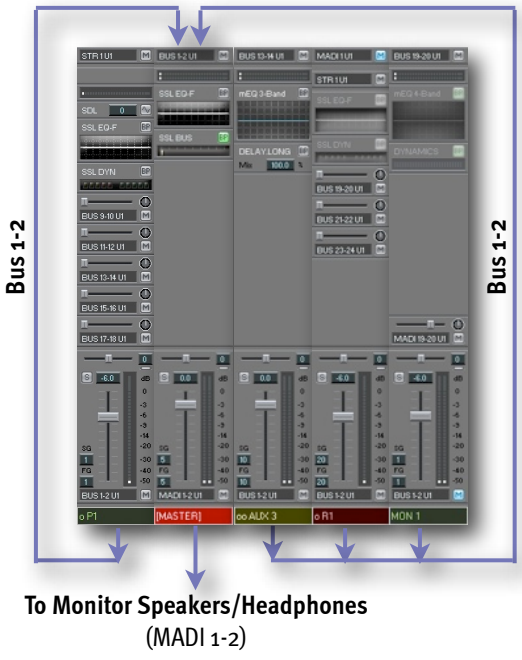
The MADI 1-2 outputs should be connected to the Control Room Monitoring speakers via the A/D/A interface, you can alternatively change the Output for example to MADI 63-64, when your Master DA with your speakers is connected there.



Master Bus Signal Flow
The Master Bus on the Input is simply fed by all other Channel Outputs using Bus 1-2.
The Output routes to a physical output, most probably directly connected to your Monitoring System(s) (Speakers/Headphones) and/or an additional "Tape" recorder.

When necessary you can change individual Channel outputs to route directly to another physical output and therefore bypass the Master Section. This could for example be useful if you want to send a clean Guitar recording to an Amp/Cabinet combo and record the results with a Microphone over a Recording Channel.

Due to the **near ZERO latency** architecture of the SSL Mixer V6, this is now all possible directly inside your PC.



5 Stereo AUX channels are set up and can be used as Sends for Reverbs, Delays, Chorus and other EFX you want to use globally.

Please note that effects inserted into these channels should be set to a 100% wet mix, since they are working in parallel with the original channels coming from the Mixing Channel Section.

The 5 Aux Channels are set to receive their Inputs from the Mixing Channel's sends on Buses 9-10 to 17-18.

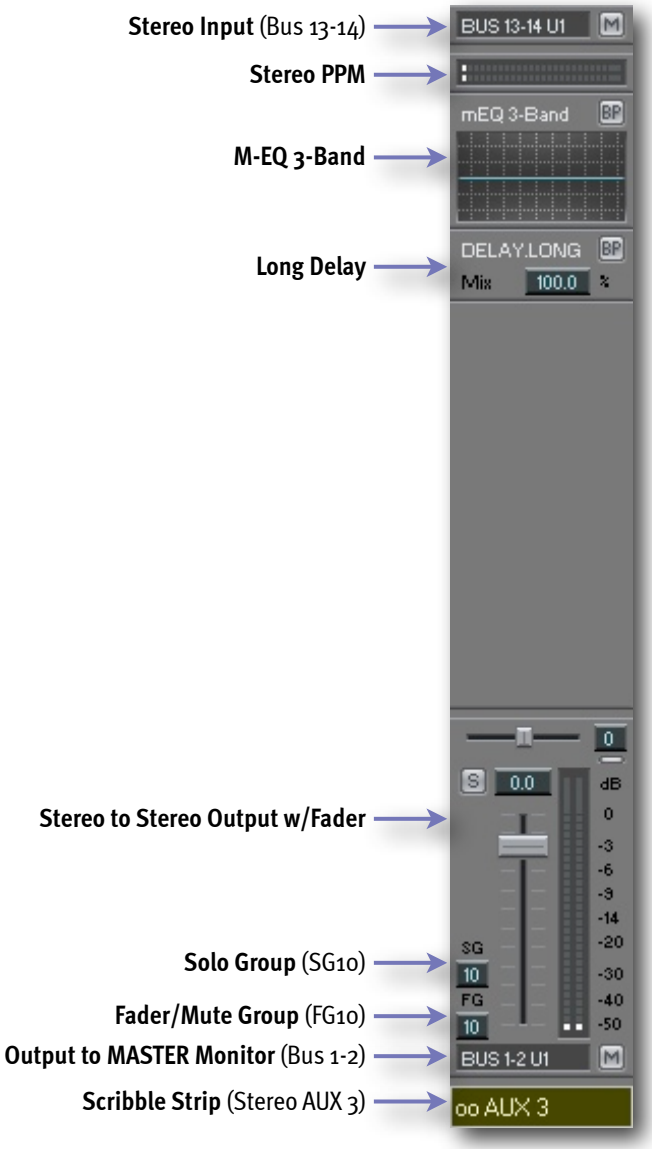
AUX 1 and AUX2 have no Plug-Ins inserted and are therefore ready to for you insert a VST or SSL Mixer DSP effect.

Aux 3 has a short slap back delay and a M-EQ inserted, Aux 4 is is ready with a psychedelic stereo ping-pong delay and also some M-EQ action.

Aux 5 has a Chorus/Flanger element inserted.

All Aux Channels are set to a common Solo and Fader Group 10, so you can easily Solo 1 effect or listen to a fully dry mix by muting all of them with a right click on a Mute button.
Also it allows you to alter the "global effect" volume by moving on of them with a right mouse click.

When you want to use an external EFX Processor, simply insert / create an Output Element, directly followed by an Input Element (both without Fader) and set the I/O's to the appropriate hardware connections.



AUX Channel Signal Flow

The AUX Channels derive their Inputs from Sends on all other Channels. We used 5 Stereo Bus pairs from Bus 9-10 to Bus 17-18.

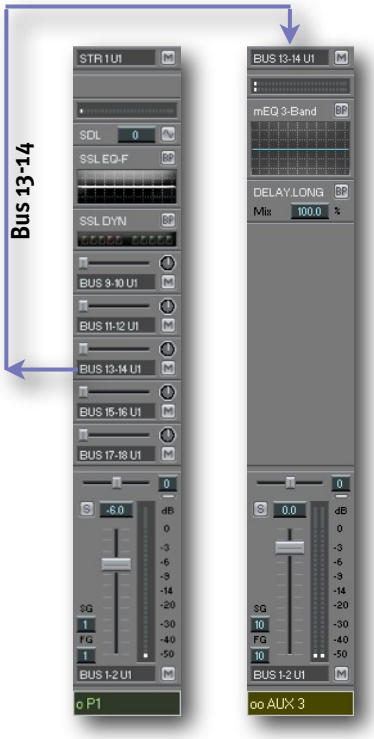
The Outputs of the AUX Channels are set to Bus 1-2, our MASTER Bus, going directly to our Summing/Monitoring MASTER Channel.

You can set up more AUX Channels if desired, and create new Send inside the other Channels to fed them. Alternatively you can change some Send Outputs in the Channels to the Bus Input of the newly created AUX Channel(s).

Again, your imagination is the limit.

Due to the **near ZERO latency** architecture of the SSL Mixer V6, creating complex Busing structures does NOT affect the overall latency of the Mixer at all.
Internally, the Buses do not introduce ANY additional latency.

However Plug-ins inside the AUX Channel can add latency, especially native VST Plug-Ins.



12 Inputs (4x Mono Ins and 4x Stereo Ins) have been set up to record 12 physical inputs (MADi 1 to MADI 12), pass them directly to Stream 1-12 into Nuendo AND down the Recording Channel Strips.

After (below) the output going to Nuendo's Stream Inputs SSL EQ and SSL Dynamics can be applied for Monitoring.
They are DSP muted and should be un-muted to use them for Monitoring.

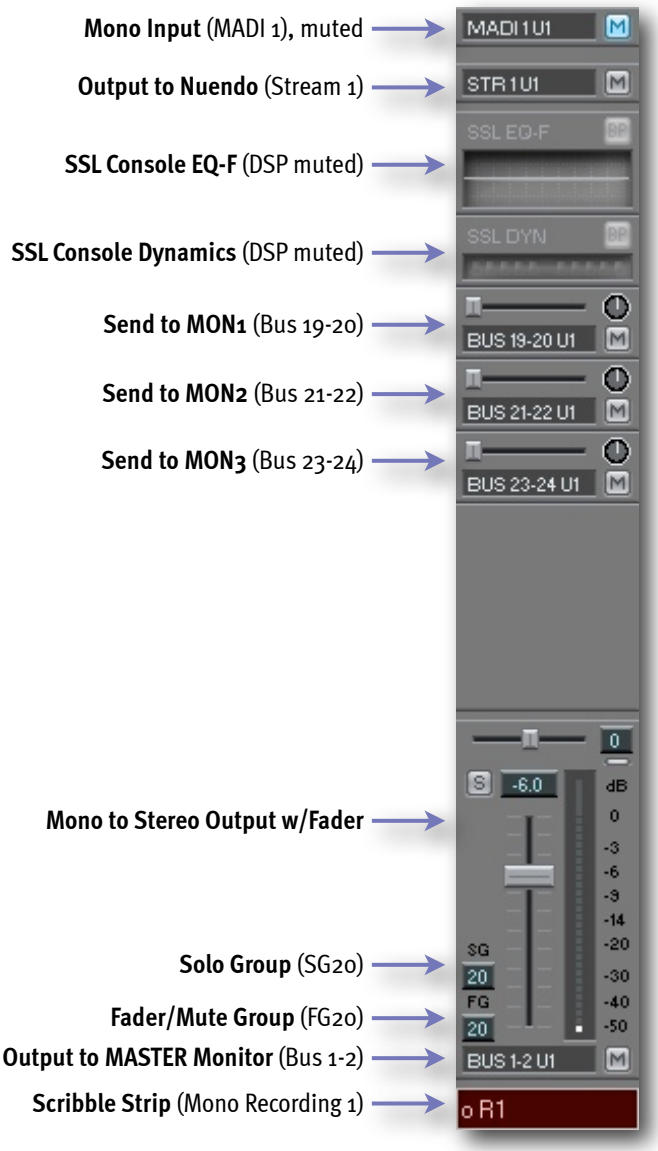
3 Stereo Sends to Monitor MON 1-3 (Post EQ and Dynamics) allow **near ZERO latency Artist Monitoring with Effects!**

The Output Section with Fader sends the Signal over Bus 1-2 to the red MASTER Strip in the mid Master and Monitor Section.

The order of elements can be changed between the top Input and the bottom Output, for example:
If the Output to Nuendo is moved below the SSL EQ and Dynamics, the signal going to Nuendo will contain EQ and Dynamics, therefore you would record the processed signal.

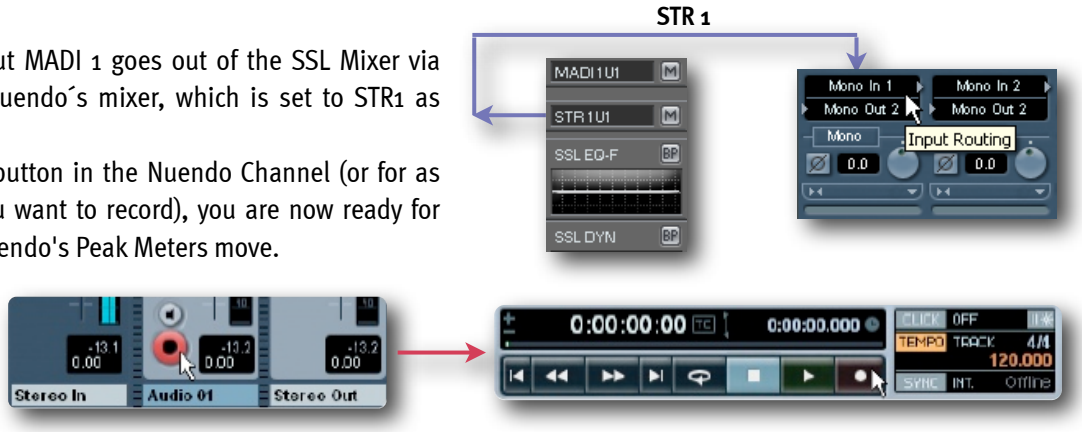
Or by moving the Send to Mon3 (Bus 23-24) above SSL DYN (below SSL EQ-F) would send an artist the EQ'd but not Comp'd signal to his headphones, the Compressor would still be audible on the MASTER Monitor, going to the Producers Control Room Monitoring Speakers.

When you use the SSL Mixer V6 for mixing only, both Recording Channels and Artist Monitor Channels can be DSP muted to free up DSP resources that are probably better used for the mixing tasks.



Recording to Nuendo

The signal coming from Input MADI 1 goes out of the SSL Mixer via STR 1 to Mono Input 1 in Nuendo´s mixer, which is set to STR1 as Input.
Pressing the record enable button in the Nuendo Channel (or for as for as many channels as you want to record), you are now ready for recording and should see Nuendo's Peak Meters move.

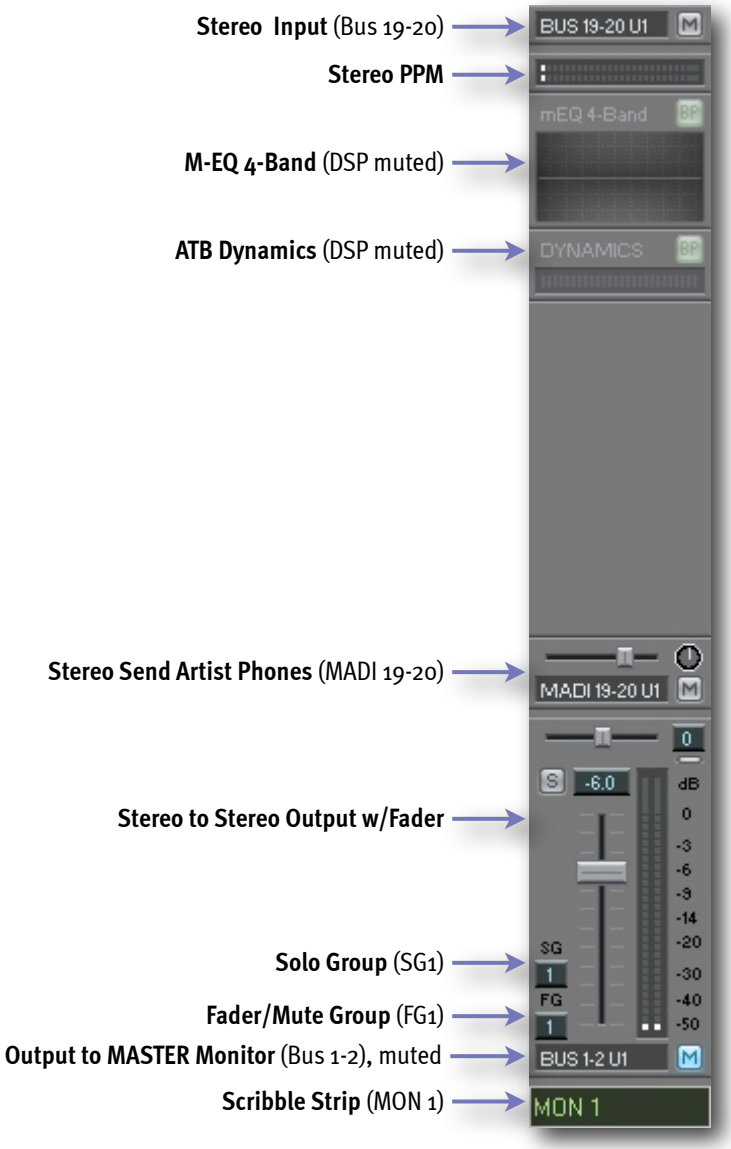


NOTE: It is also possible to use the high quality SSL EQ and Dynamics for recording, as you may do with "real" consoles without added latency.
Just move the streaming output element below the SSL EQ and Dynamics elements, so the "Tape Send" is Post Effects now.

Three stereo Artist's Monitors (MON1 to MON3) derive their Input signals from auxiliary sends on the Recording Channel.
Send Elements (Pre-Fader) routed to the MON Channels are assigned to physical Outputs (here MADI 19-20) that are ultimately connected to Headphone Preamps.
The Artist Monitor section bypasses Nuendo entirely, directly sending the input signal to the Artist with near ZERO latency. The DAW's build in Direct Monitoring should not be activated, to avoid listening to the REC Input twice.

The (currently DSP muted) M-EQ and ATB Dynamics can be used to adjust the Headphone Sound.
The Monitoring volume for the Artist is handled by the little Fader on the Send (MADI 19-20 in this case).
If MADI 23-24 (or MADI 47-48) is used, you can connect a headphone directly to an Alpha-Link headphone output, which is "hard-wired" to DA 23-24.

The channel's output level to the Master Channel is adjusted with the bottom Fader, which does not affect the Send to the Artist headphones (set to Pre-Fader). To avoid double signals to the MASTER, the Output is currently muted. Fader and Solo group is set to Group 1, to solo the MON's while muting the Mixing Channels.

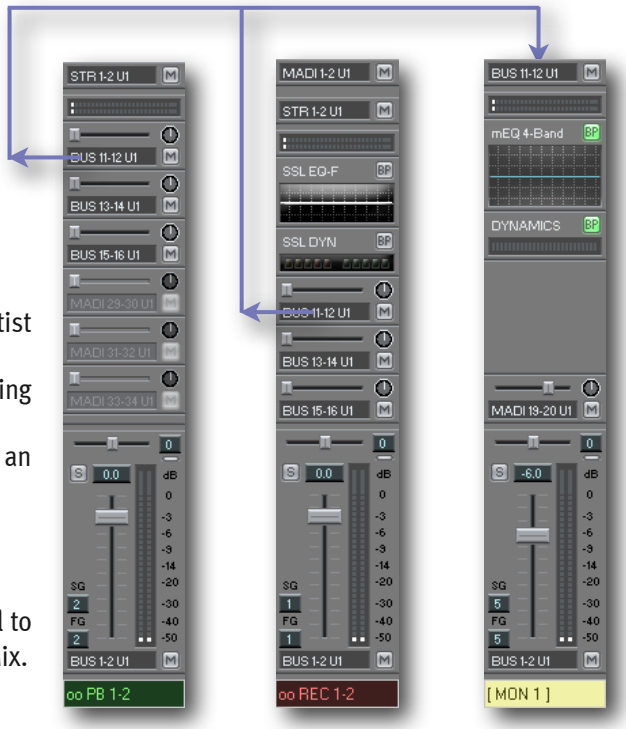


Setting up Artist Monitoring
Recording Channel
- On the armed Recording Channel locate the first Send (**BUS 19-20**) and open the fader to approx 80%.
- You should now be able to see the top PPM work in the **MON1 Channel** and hear the Recording Channel's Input through your Monitoring System

MON1 Channel
- Open the Send (MADI 19-20) in the Artist Monitor Channel MON1
- You should now be able to hear the Input from the Recording Channel on the headphones.

Playback Channels
- All Playback Channels, that should also be fed into the Artist Headphone, also need a Send assigned to Bus 19-20.
- Simply create a new send or assign Output to Bus 19-20 on existing Sends.
- You can now feed every individual Playback Channel with an individual send level and pan into the Artist Monitor.

Pre Listen the Artists Headphone Mix
- Solo the Artist MON1.
- Click the Mute Button on the Output of MON1, to send this channel to your MASTER Channel. You are now listening to the Artist Monitor Mix.



The SSL Mixer runs entirely on the DSP chips of the MX4 and 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 in the SSL 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".

The SSL Mixer also supports full automation. It can generate and respond to MIDI automation data via the Console Manager program. This data must be recorded to a third party application (MIDI sequencer).

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, you can have up to 128 mixer strips 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 sequencing application.

Loading and Saving a MIX

By default the "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 will be loaded on next startup.

"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).

4 Mono and 2 Stereo mixer strips are prepared for recording. 5 Stereo Send Buses and 3 Stereo Artist Monitors complete the Mixer and allow overdubs with **near ZERO latency Monitoring** and global EFX.

To use this Mixer, outputs 1/2 of your SSL hardware must be connected to your monitoring system via a suitable interface. Windows applications will also need to be configured to use the SSL audio streams according to its accompanying documentation.
The SSL audio streams have names like for instance "STR 1-2" to "STR 127-128" in the Windows Control Panel (for stereo pairs), and the name of the card is also specified.

You can load different mixers to experiment with by clicking "Open Mix" under the File menu.
A standard Windows dialog box will appear allowing you to select a mixer.
The default path for mixer files is: C:\Soundscape\Mix\Mixer

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 sequencer 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, in a 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

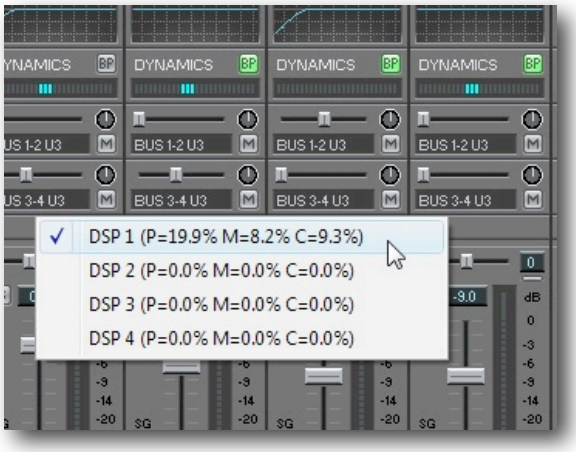
Assigning a mixer strip to a specific DSP

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 for the left mouse button by clicking on its icon: 

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.

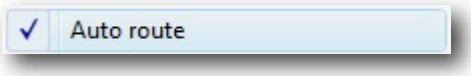
The ticked entry indicates the DSP Core that the mixer strip is currently assigned to (DSP 1 in this example). The DSP assignment can be changed by chosing another DSP before releasing the mouse button.



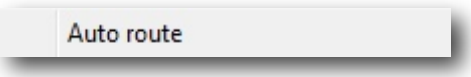
Mixer auto routing

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.

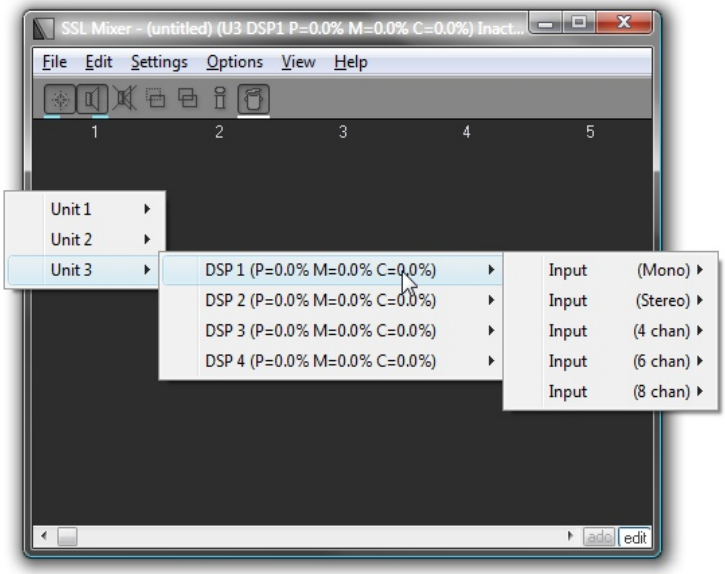


Mixer auto routing disabled
If the "Auto route" item under menu: Settings|Preferences|Mixer is **unticked** (auto routing disabled), the DSP resources need to be allocated manually when building a Mixer, as described below.



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).
A submenu then lists all the input element types, a further submenu will list the output element types.



Basic Mixer Strip Structure

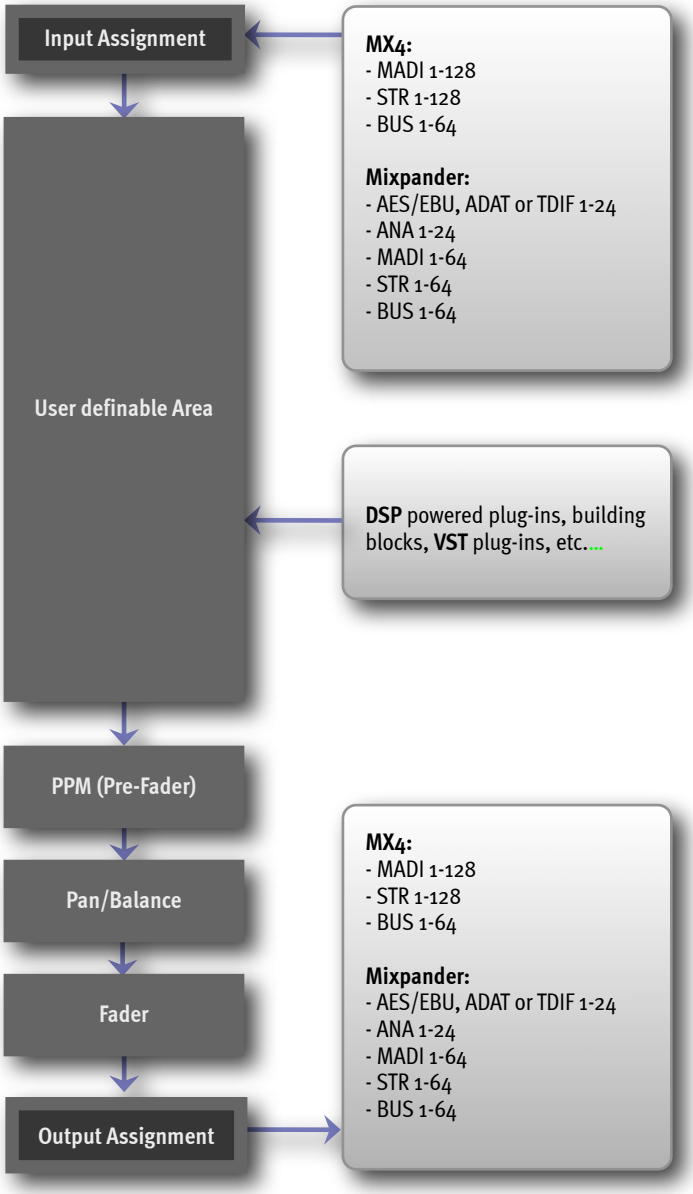
This diagram shows signal routing through a stereo input mixer strip.

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.

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

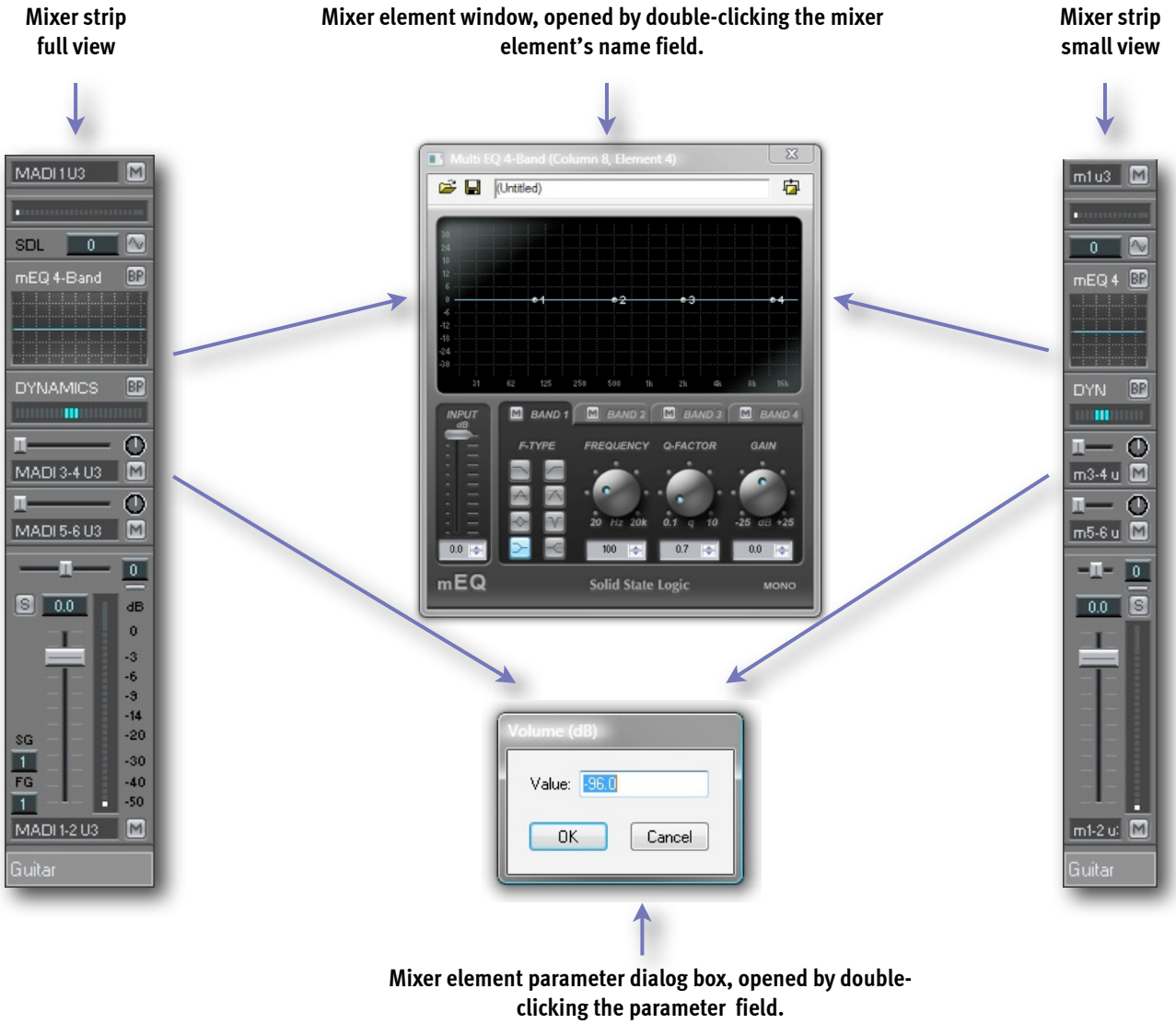
The SSL Mixer can be displayed in one of two modes: full column size (wide) or small column size (narrow). Both options are available under the View menu or by pressing the [X] or [Q] key on the computer keyboard.

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.

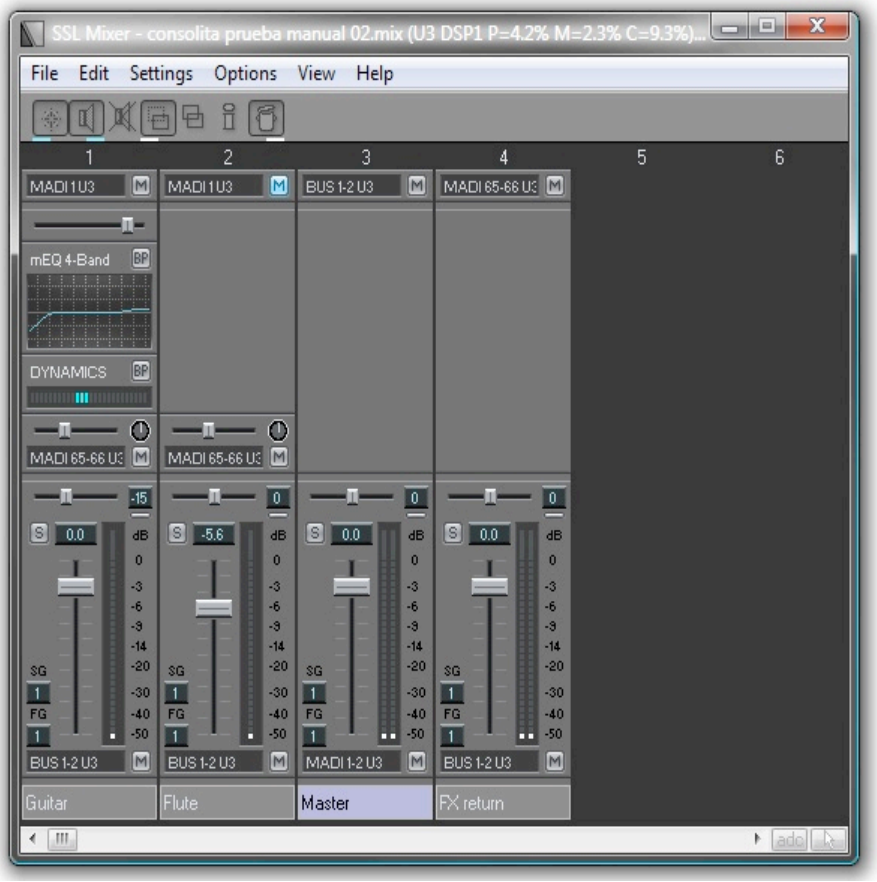
The Mixer contains **128 mixer columns,** each contains one mixer strip that is 1 to 8 Channels "wide".
A horizontal scroll bar is provided for navigation across the Mixer window.

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 any PC DAW user.

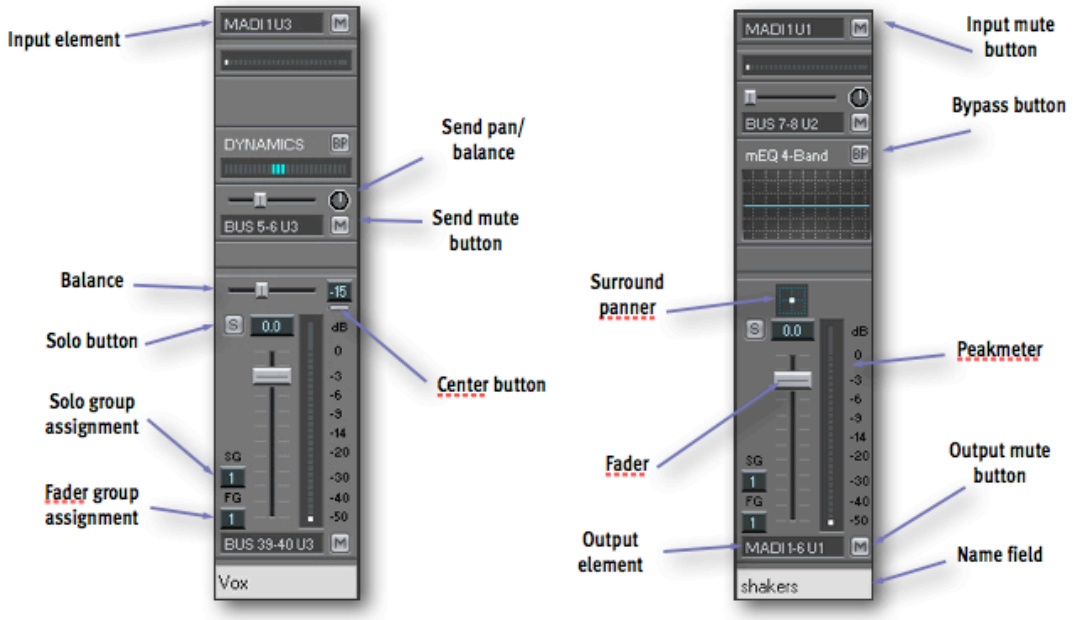
For example, if you want to send an EQ'd signal to an effects unit, drop an EQ in before the send. If you want a flat signal sent to the effects unit but want an EQ'd version sent to the master out, drop an EQ in after the send. If you need one auxiliary send in a strip and eight in another strip, just create them as required!



Mixer strips

Mixer Strips in columns have the following standard controls and buttons:

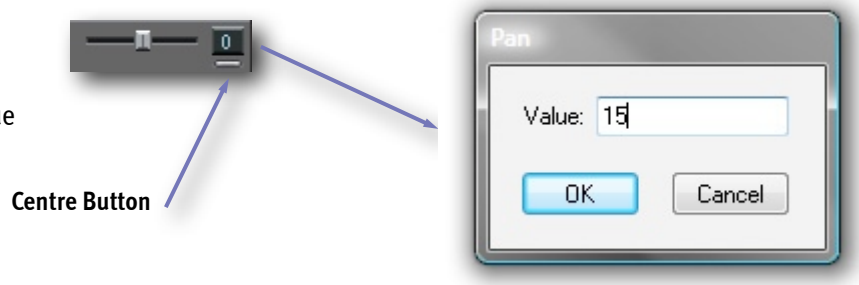
- **Input element** on the top and an **output element** on the bottom, each with **mute buttons**
- **Scribble Strip** (name field) with definable Font and Background Colour)
- On **Outputs with Fader**: a pre-fader peak meter, a solo button, Volume Fader, a solo group assignment box, and a fader and 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



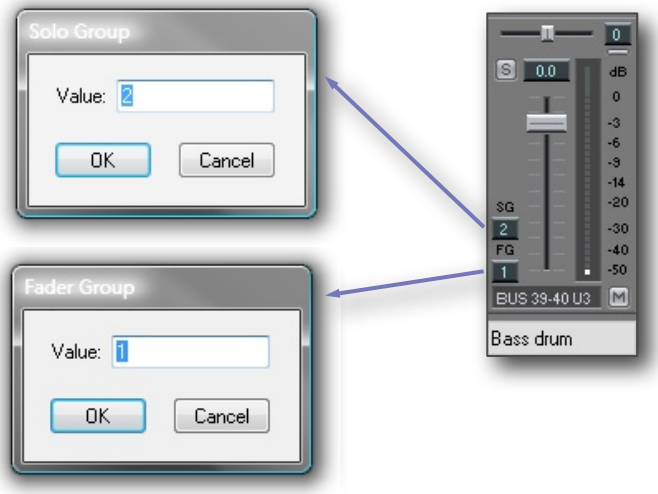
Fader and mute groups

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 group.

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.

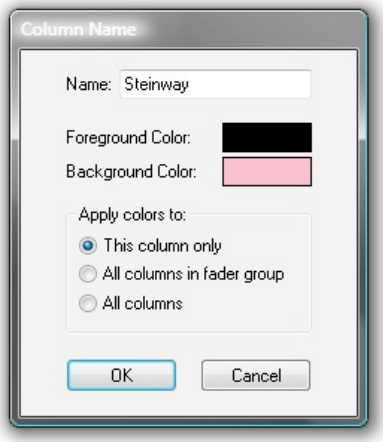
- **Right-clicking** an inactive solo button **solos the corresponding output and all the outputs that have the same fader and mute group assignment.**
- It also **mutes all the outputs** that have both a **different fader and mute group assignment and the same solo group assignment** as any one of the 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 a 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 Mixer 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.

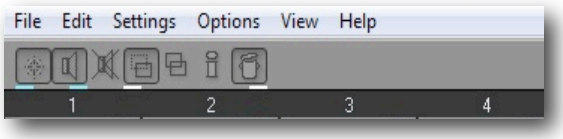
NOTE: When working at 96Khz, the number of connections to the Alpha-Link Unit and Streaming I/O is reduced to 32.

Soundscape 32 workstation

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 Edit Mode Toolbar



The SSL Mixer 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 could dream of.

NOTE: The Tools only work, when your 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 white 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 light blue 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

Copy tool: Copy mixer elements inside Strip or mixer strips to another column

Delete tool: Delete mixer elements or mixer columns

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

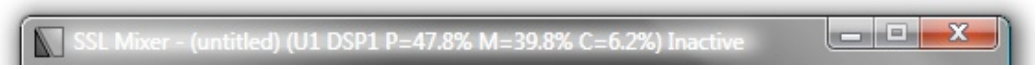
Each Tool's function is explained in detail in Chapter 9. Menu Reference, Section Edit Menu.

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" and "M" values, and the changes you make will be reflected instantly on screen.

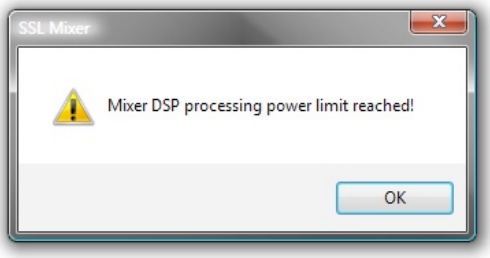


The [Shift] key can be released before the last edit is carried out in order to reactivate the edited Mixer.

Alternatively you can double click on the Caption bar to reactivate the Mixer.

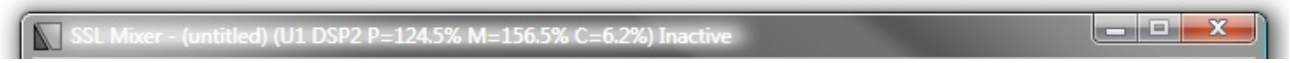
DSP power limit

The "P" value reported in the Mixer window's title bar is the percentage of the available DSP clock cycles which is 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:

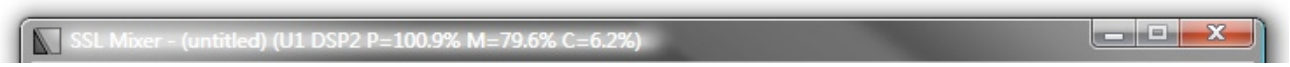


After you click "OK", the title bar will display the P and M values so that you can see which one exceeds the available resources, and by how much. If only the "P" value exceeds 100%, the title bar will display a warning as shown below: [create mixer 05.bmp]

If the "M" value or both values exceed 100%, the title bar will inform you that the Mixer is inactive: [create mixer 06.bmp]



Muting or deleting some elements will reduce the values and reactivate the Mixer.

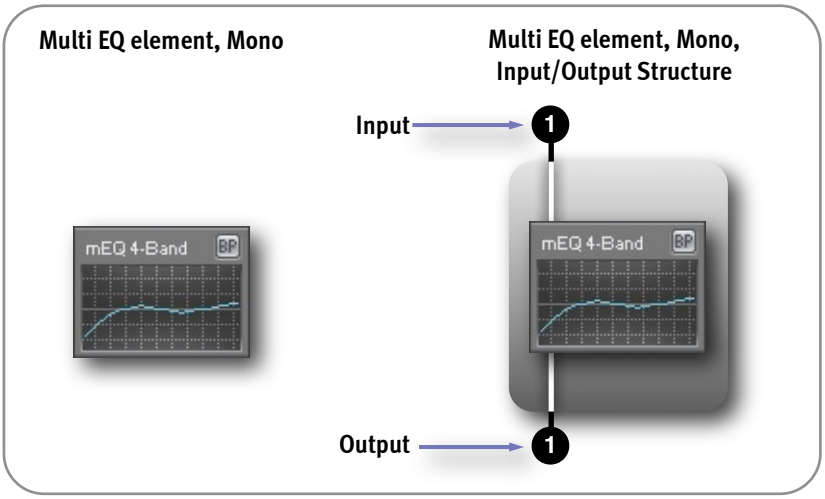


However, the P=xx% value is not completely accurate, because some plug-in elements report inaccurate processing cycles to the SSL Mixer. This means that depending on what kind of elements are used, some Mixers can't reach or may exceed the P=100% value without the warning box appearing.
Also the Mixer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

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 "recorder"**. It is helpful to visualize them in order to understand the signal flow within a mixer strip.

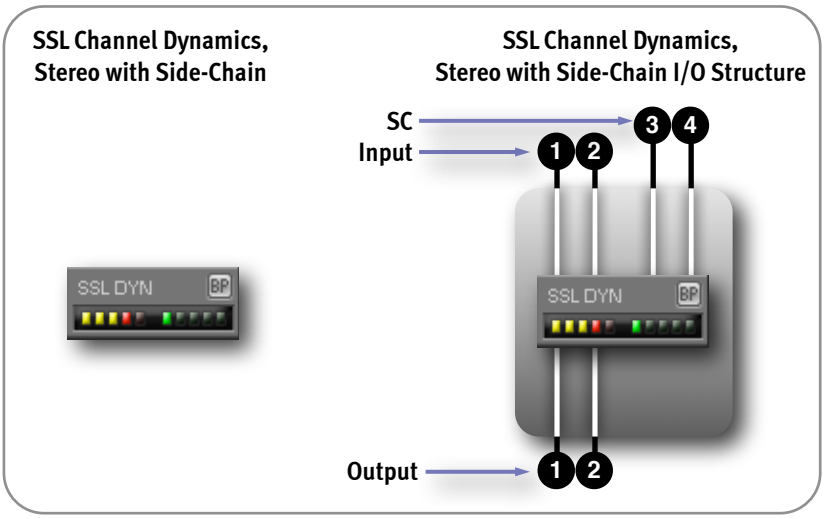
Fixed inputs/outputs

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.



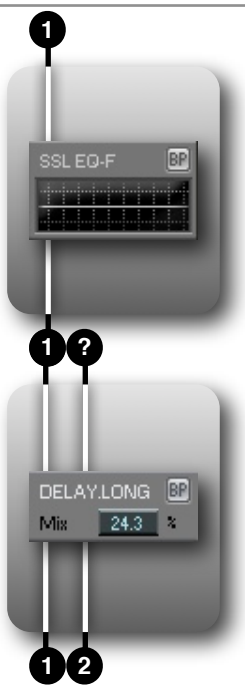
...and a stereo SSL Dynamics element with side-chain, as it appears within a mixer strip, and with its four inputs and two outputs represented.

Fixed inputs and outputs always connect to an output or input with a matching number. 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:



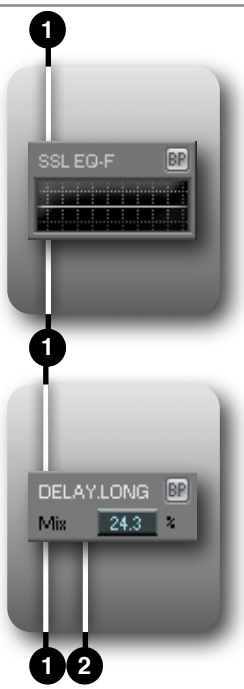
Incorrect configuration:

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 routings.



Correct configuration:

A mono mixer element (SSL Console EQ Mono), is connected to a mono to stereo mixer element (Audio Toolbox Delay Mono->Stereo)



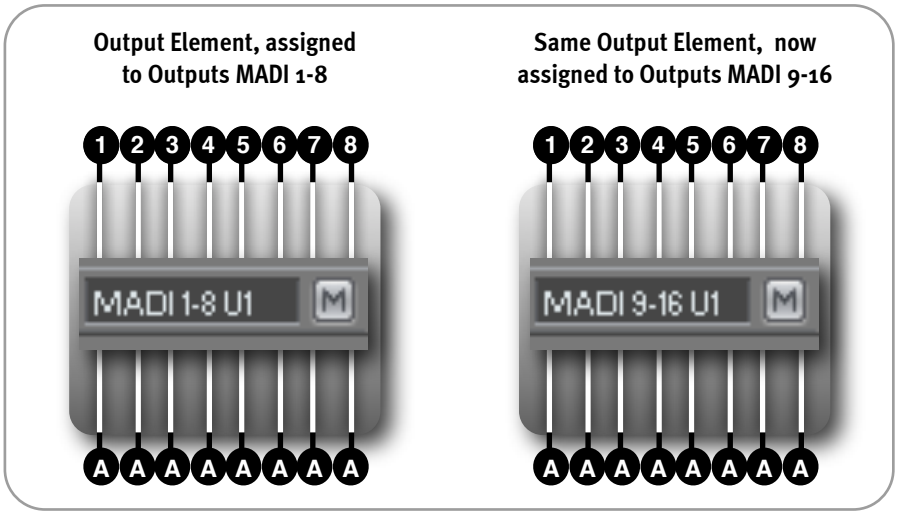
It is important to take this into account when building a Mixer, since the two configurations shown above would look the same within a mixer strip. If you need to check the input/output configuration of an existing mixer element, click on it with the Info tool in Mixer Edit mode. A window containing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

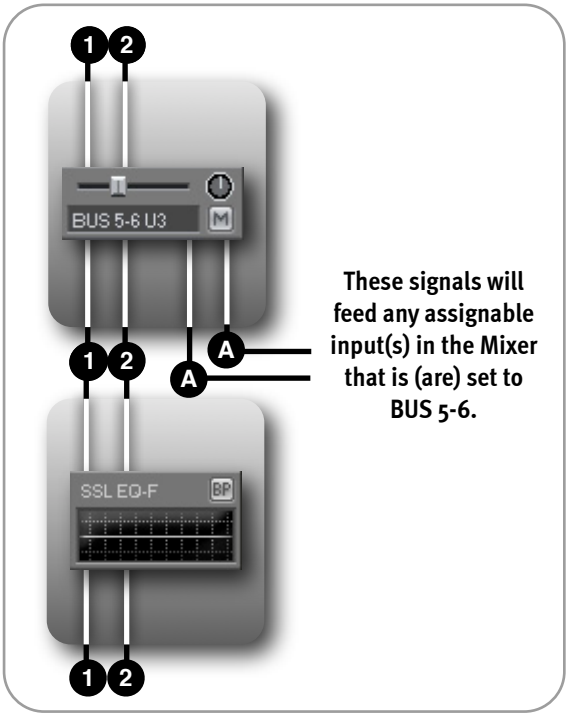


The example above shows an 8 in/8 out output element. Initially, the element's outputs are assigned to outputs 1-8. The I/O Assign tool is then used to change the element's output assignment to outputs 9-16.

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



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 up a signal from elsewhere in the Mixer).

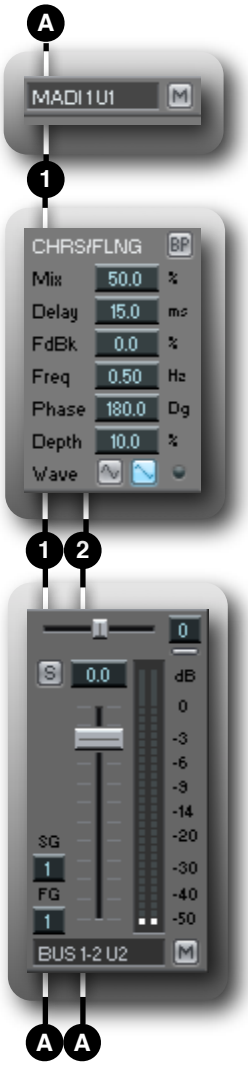
In the example on the right side, a mono signal reaches the mixer column via the MADI 1 input.

A mono to stereo Chorus effect is applied.

The Chorus stereo pair of outputs connects as required to the stereo output element.

mono in - stereo out mixer strip as seen in the Mixer Software

mono in - stereo out mixer strip connection diagram



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

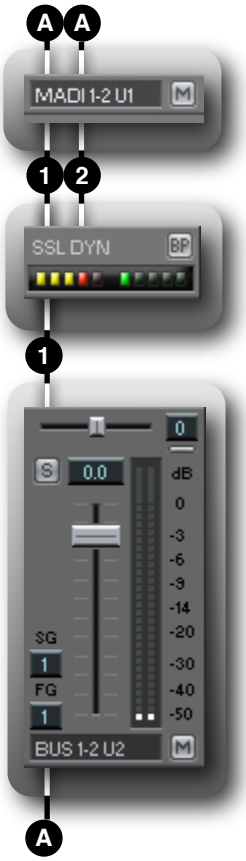
This configuration can be used to insert a 2 in/ 1 out mixer element such as a mono compressor with side-chain input.

The main signal will reach the compressor via MADI input 1, the key signal will reach the compressor via MADI input 2.

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

stereo in - mono out mixer strip as seen in the Mixer Software

stereo in - mono out mixer strip connection diagram



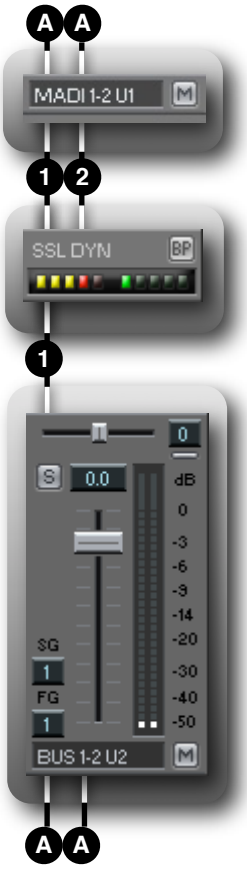
For stereo in - mono to stereo out, the left side of the stereo input is connected through the mixer column, but it splits into stereo at the pan control.

This configuration, just like the stereo in - mono out configuration described above, can be used to insert a 2 in/1 out mixer element such as a mono compressor with side-chain input.

The only difference is that the output can be panned left or right as required with the mono to stereo output element.

stereo in - mono to stereo out as seen in the Mixer Software

stereo in - mono to stereo out connection diagram



Stereo Balance and Panning

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

4 in - stereo out mixer columns

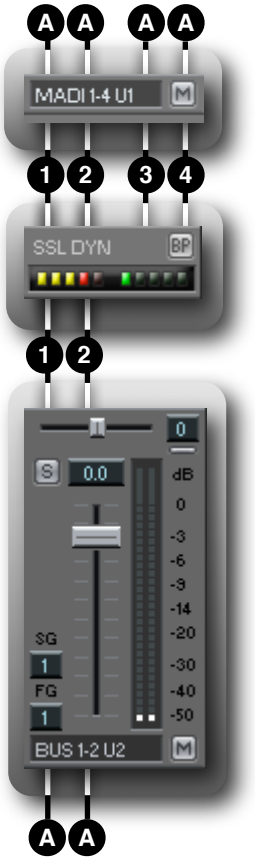
This configuration would be used to insert a 4 in/2 out mixer element such as a **stereo compressor with side-chain** input.

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.

4 in - stereo out as seen in the Mixer Software

4 in - stereo out connection diagram



Surround Mixing I/O Configurations

For mono in - mono to LRCS out, the signal path through the mixer column is mono, allowing mono processes or effects to be applied to the signal.

The mono multi EQ in the example, is applied before the eq'd signal is positioned in the sound field using the **surround panner**.

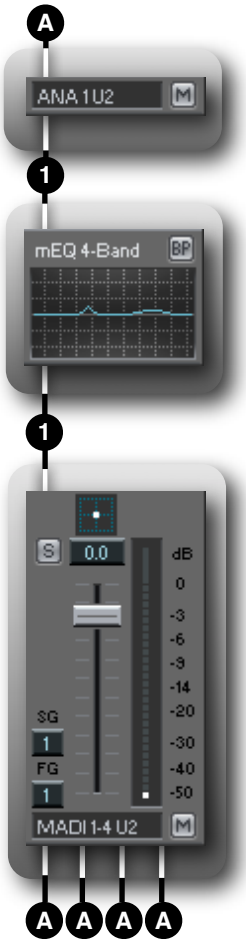
A **stereo in - stereo to LRCS out** configuration is similar to the mono in- mono to LRCS out configuration, except that the signal path through the mixer column is stereo, allowing stereo processes or effects to be applied to the signal.

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

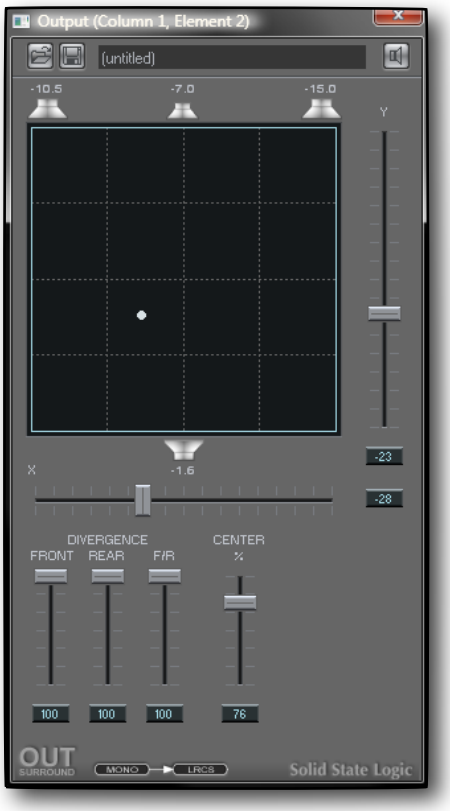
- MADI 1: L
- MADI 2: R
- MADI 3: C
- MADI 4: S

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

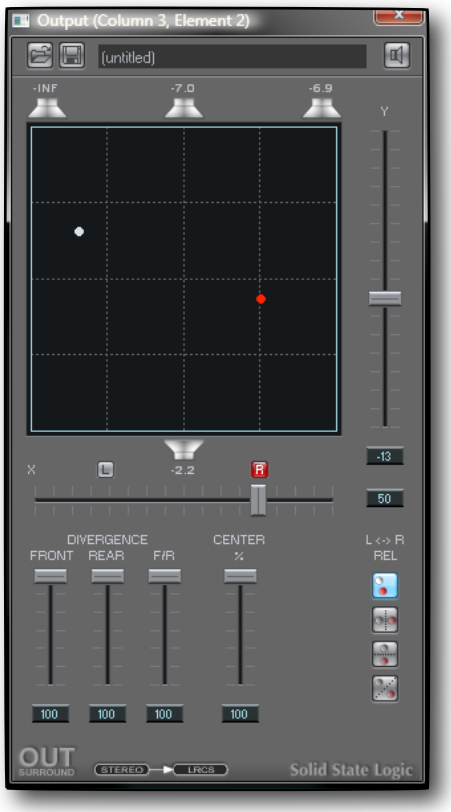
mono in - mono to LRCS strip connection diagram



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

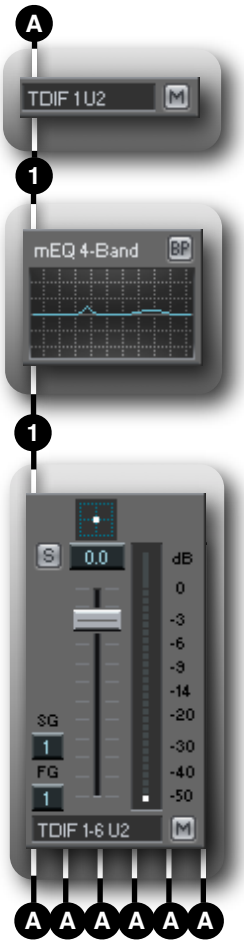
The **stereo in - stereo to 5.1 out** configuration is similar to the mono in- mono to 5.1 out configuration, except that the signal path through the mixer column is stereo, allowing **stereo processes** or effects to be applied to the signal.

Routing in this example:

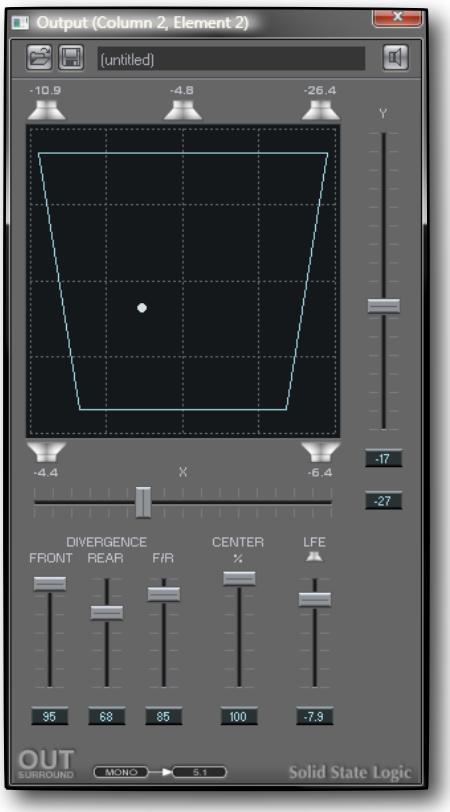
- MADI 1: L
- MADI 2: R
- MADI 3: C
- MADI 4: LFE
- MADI 5: L S
- MADI 6: R S

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

mono in - mono to 5.1 strip connection diagram



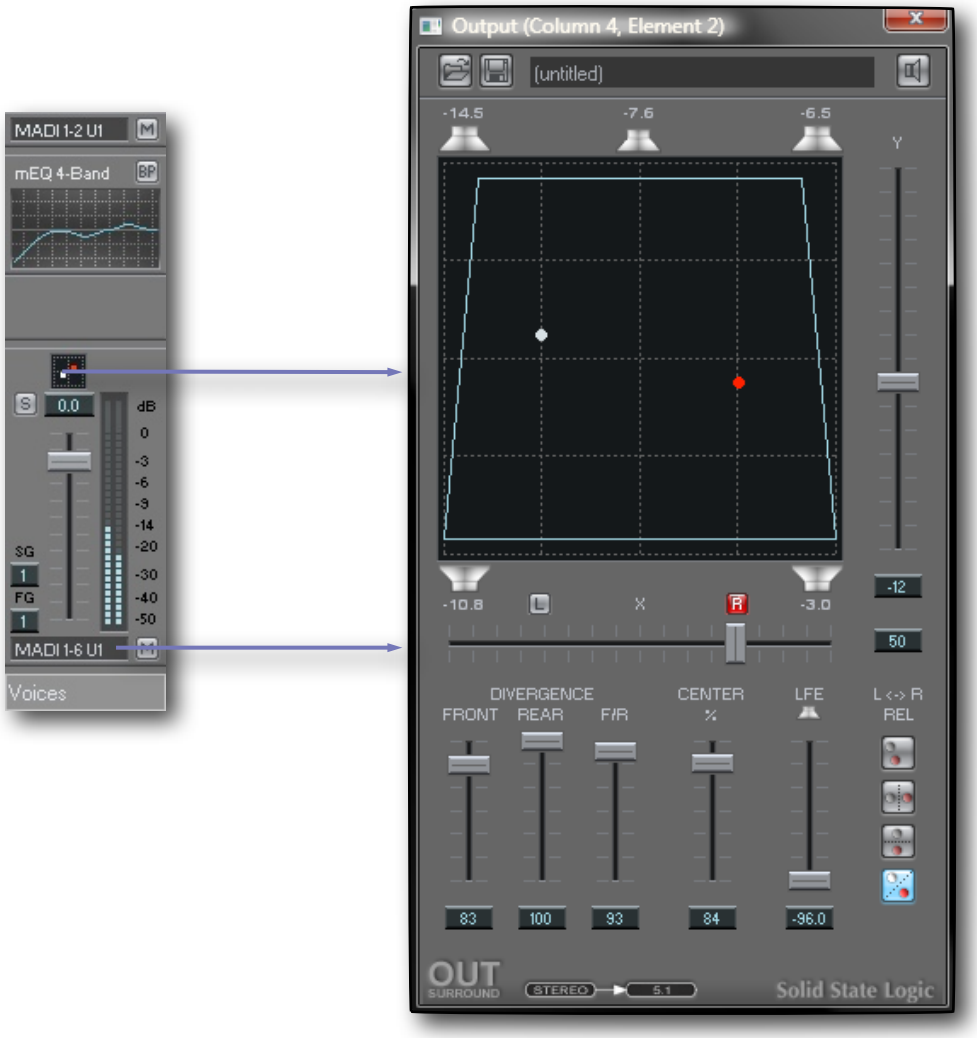
mono to 5.1 panner



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 the all 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:

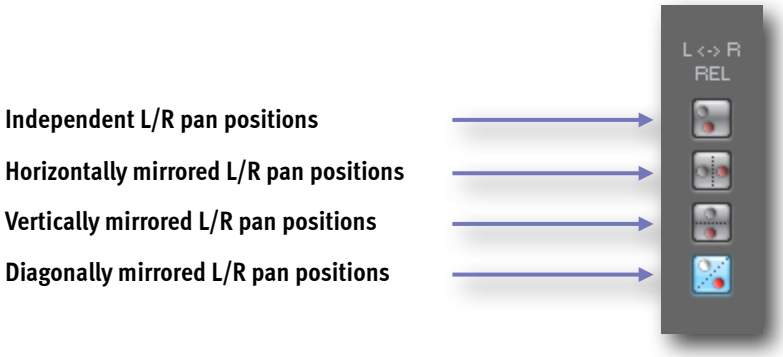
- The sliders can be moved either by clicking and dragging them or by clicking at the required position with the mouse
- Right-clicking or left-clicking in a value box respectively increases or decreases the corresponding value in steps of 1. Right-clicking or left-clicking in the selection box and holding 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!)**:**

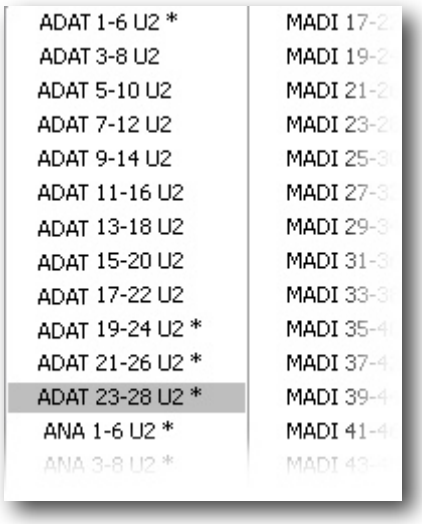
When using a Mixpander card the component signals of a surround mix may be output via 8 channel ADAT of TDIF connectors. This way, the whole mix could for example be sent to an 8 track DTRS machine for mastering (a 5.1 mix could be output via the first six outputs, along with a stereo mix via the last 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

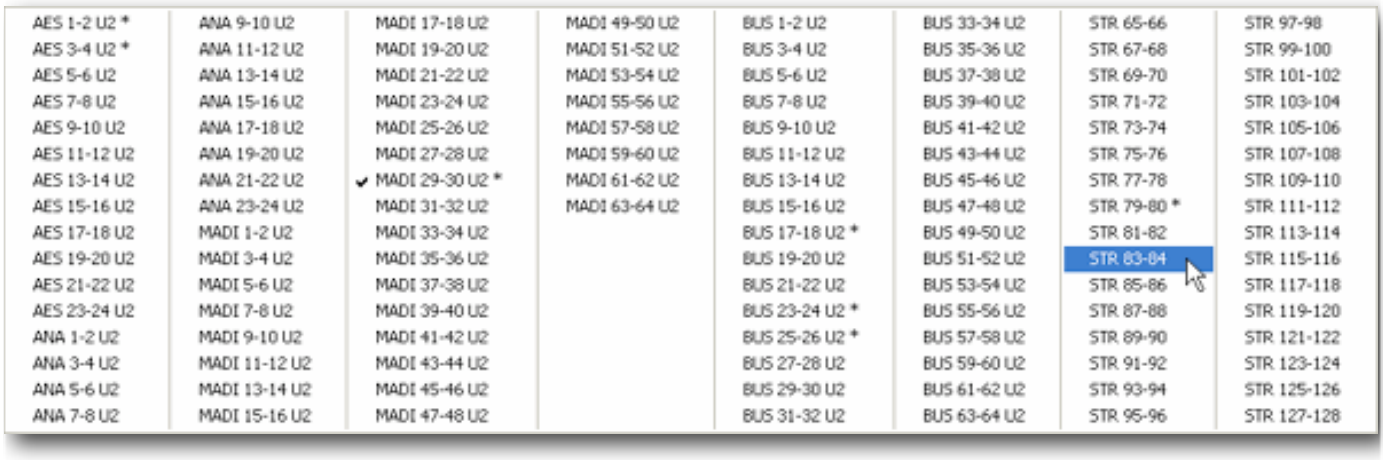
In addition to all what has been mentioned above, mixer columns can also be created with 4, 6 or 8 inputs and outputs. These could be used for example to send and receive multitrack submixes to and from 8 Channel (ADAT/TDIF) ports.

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 a **SSL Alpha-Link MADI-SX** connected to the Mixpander that the mixer column is running on.

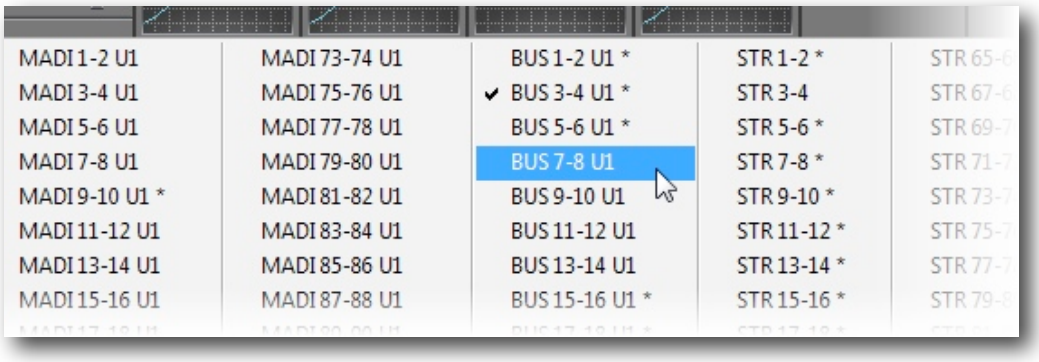


NOTE: The inputs and outputs, buses and streams that are already in use are indicated by an asterisk: (*).

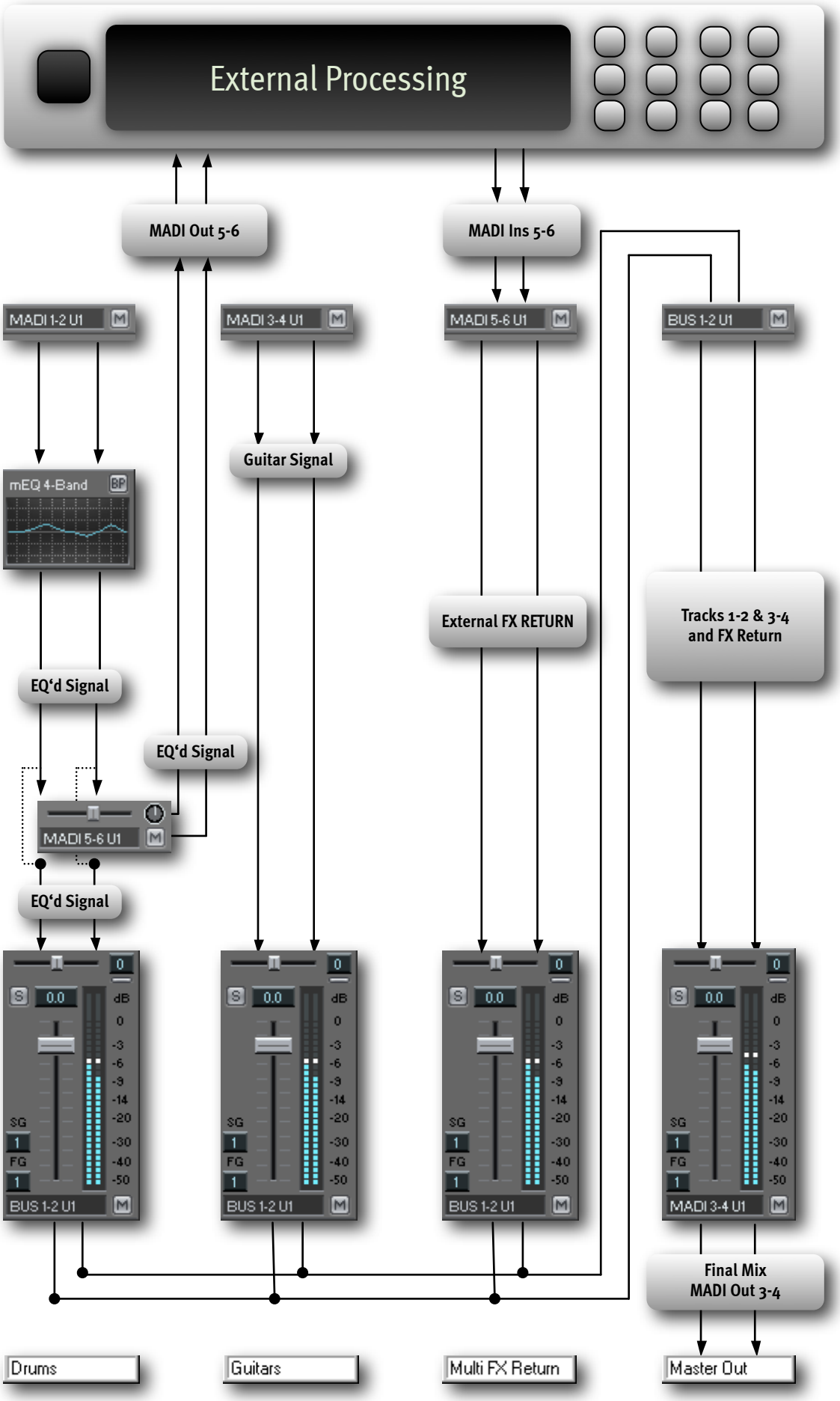
There are no restrictions to how you can use the inputs, outputs, buses or streams, and every signal routed to an output or bus is mixed in equal proportion (after the output or send fader). For instance, you could have 8 channels with MADI 1-2 U1 (or Bus 1-2) as the input, and Bus 3-4 as the output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



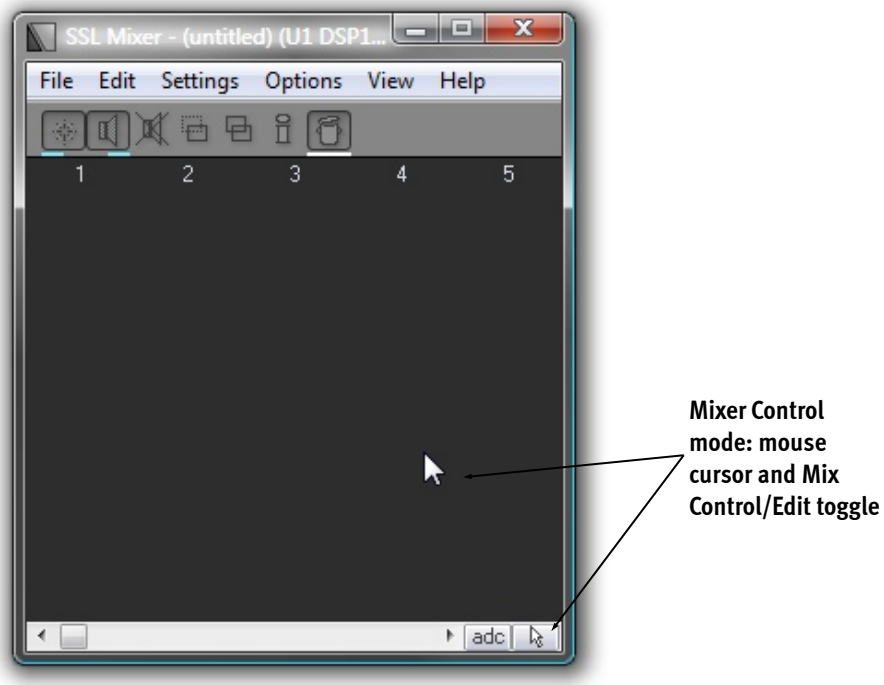
NOTE: If the output(s) from several mixer columns are sent directly to an external output (or group of outputs), it is not possible to view the combined output signal in order to check for overload. Therefore, it is advisable instead to route the mixer column outputs to a bus, and then to create an additional 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
Guitar Signal
mEQ 4-Band
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

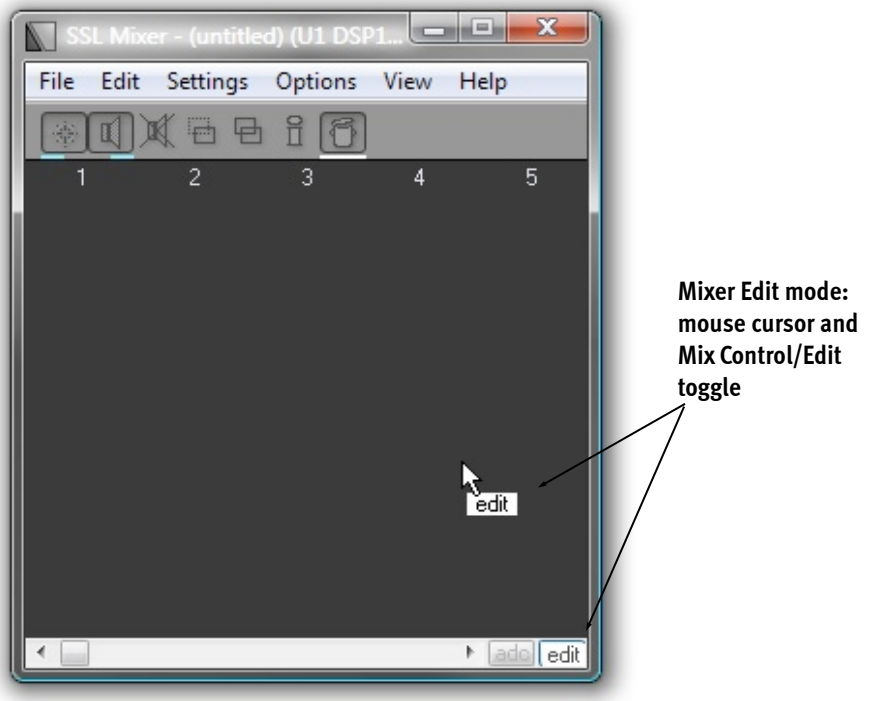
Creating a new Mixer

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" 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:

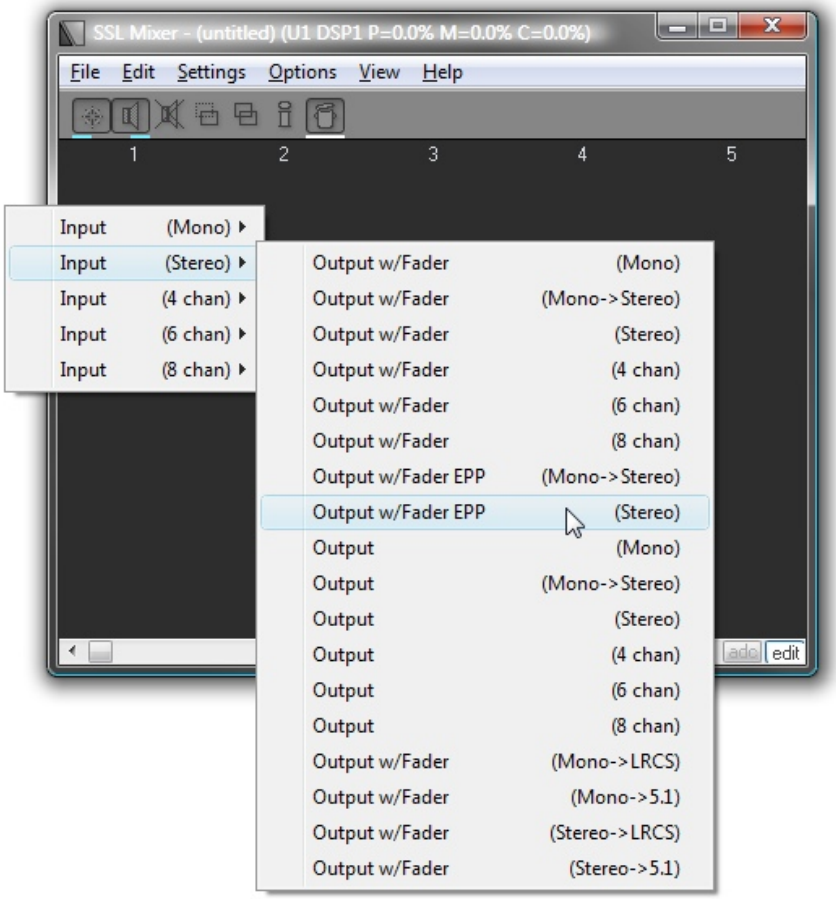


Now press the **Edit toggle button** in Mixer window or press [E], now in Mixer Edit mode the mouse cursor shows "edit" as well as the Edit toggle button:



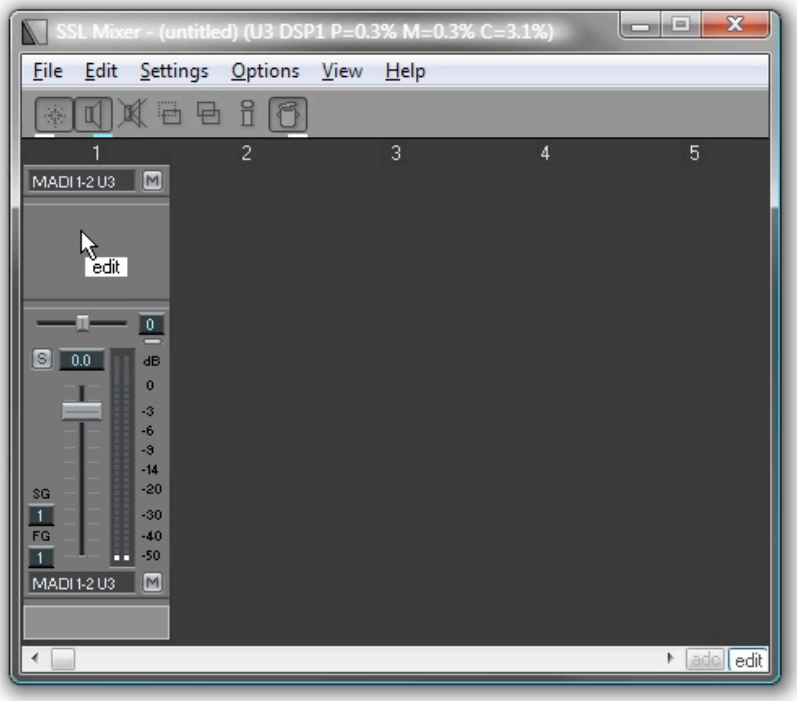
Select the **Create tool** by clicking its icon with the mouse.

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.

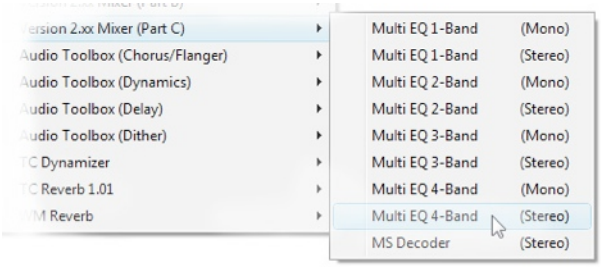


A **mixer strip** will appear with the default inputs and outputs (1-2 U1), a balance control slider with centre button, a pre-fader stereo peak meter, input and output mute buttons, a solo button and a fader.

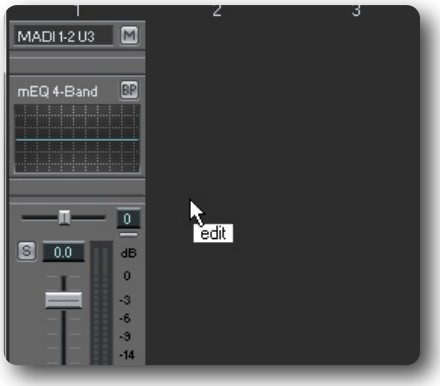
There is also a blank space at the bottom of the mixer strip that can be used to name and colour code the Strip (Scribble Strips).



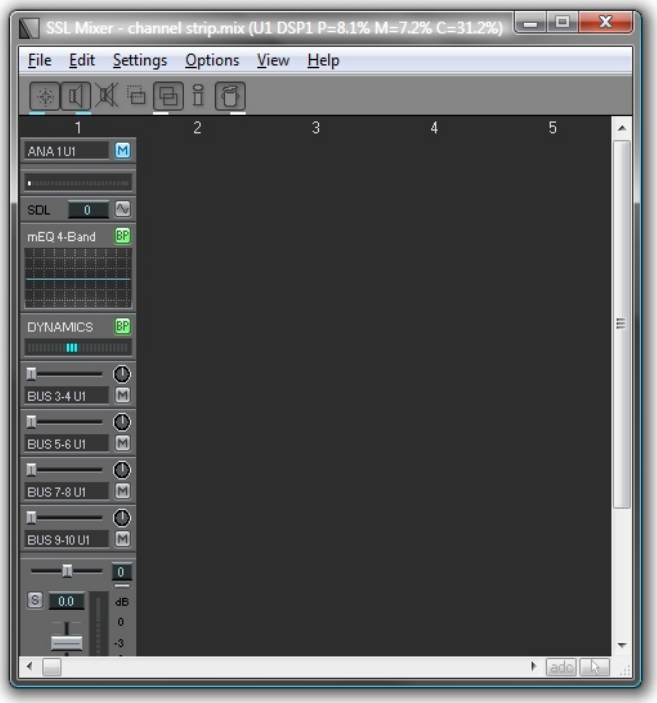
Next, **select the Create tool again,** click in the blank area of the mixer strip under the track insert. The mixer elements menu that you used to create the track insert will appear again.
This time, select "multi EQ 4-Band (Stereo)", then release the mouse button.



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.



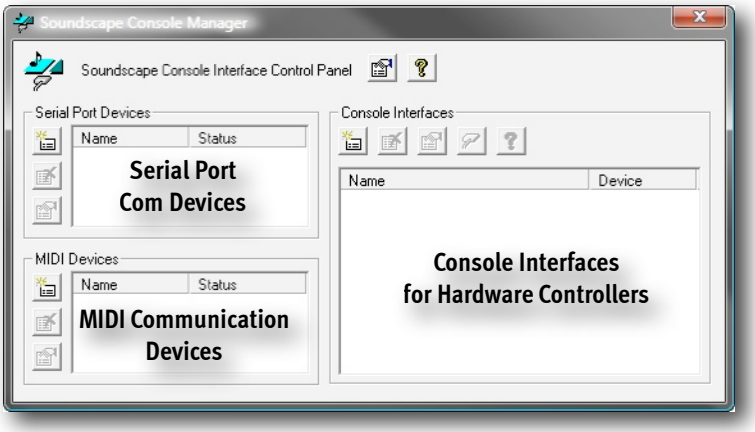
Signal routing in the mixer strip
The signal routing is always from the top to the bottom of a mixer strip. For instance, if you place a track insert below an EQ, this is the position at which audio will be recorded or played back (i.e., if an audio sequencer records from this particular track insert,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

Console Manager Properties Window

Click on Properties or double click the task bar icon opens the main Console Manager Properties Window

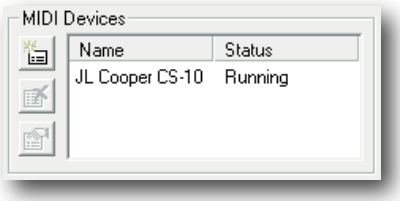
Console Manager allows external devices to control the SSL Mixer software by receiving data from these external devices and transmitting it to the Mixer Software, using several possible Control Protocols (HUI, MCU, JL Cooper etc.) that can be set up in the **Console Interface section** on the right hand 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.).

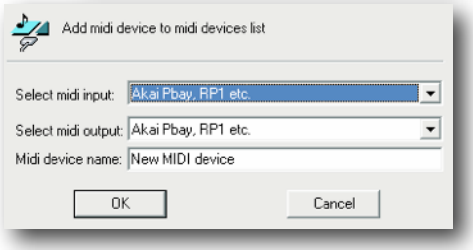
Setting Up Midi Controller Devices

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.



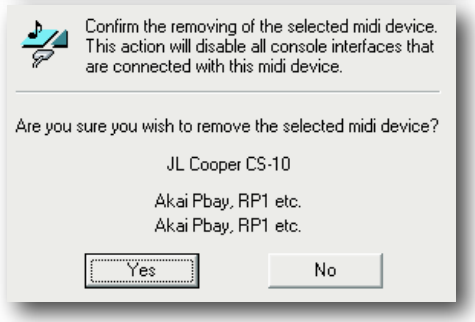
Add button

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 ‚good' 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 aborts the operation.

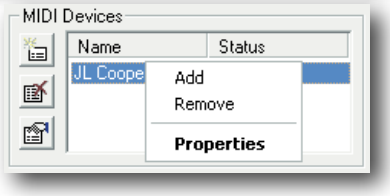


Properties button

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.

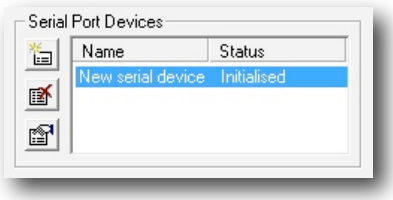
Right Click Menu

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



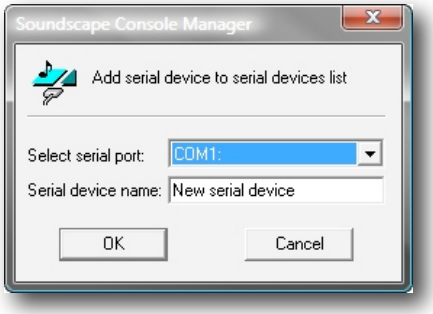
Setting Up Serial Com Controller Devices

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.



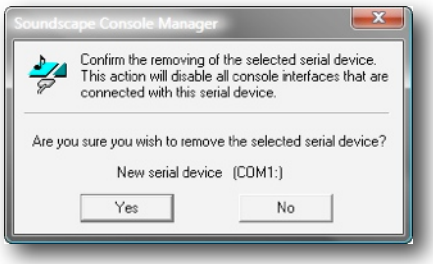
Add button

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 ‚good' name you can remember.



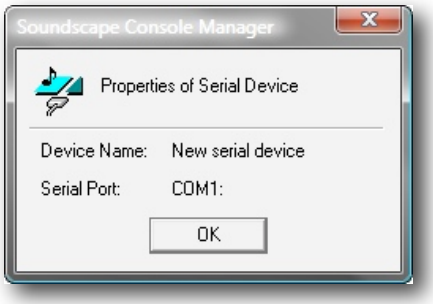
Remove button

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 aborts the operation.



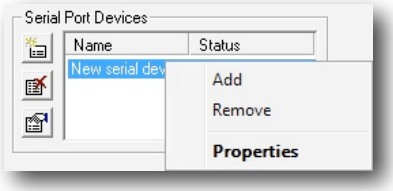
Properties button

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.



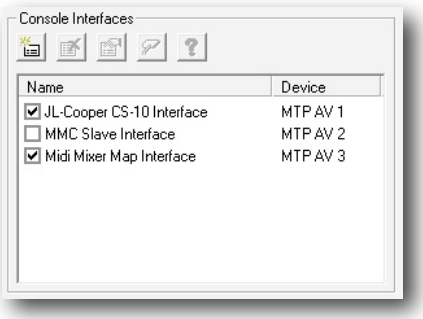
Right-Click Menu

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

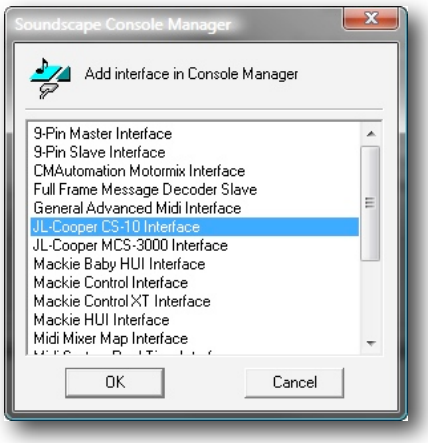


Setting up Console Interfaces

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.



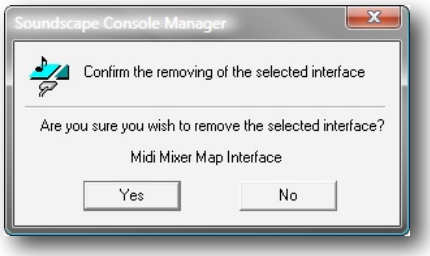
Add button



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.

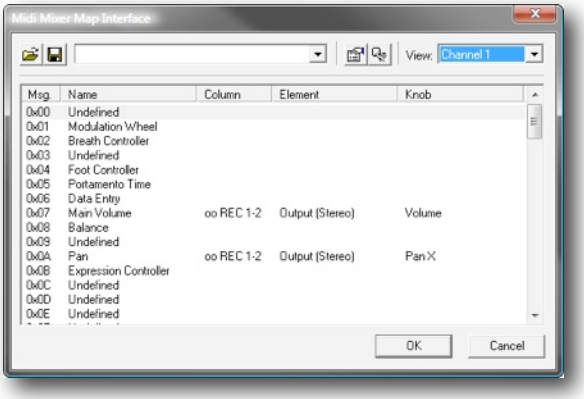
Select the appropriate Protocol/Device Interface and press **OK.**

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.

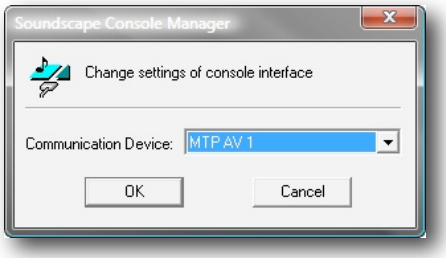
Properties button



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



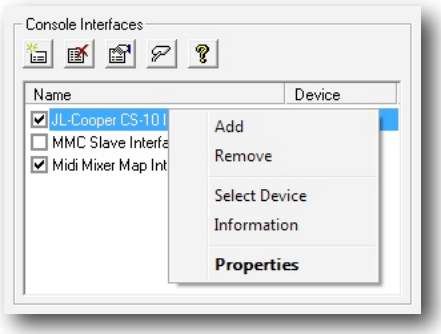
Clicking this button when a Console Interface is selected opens a dialogue box where any of the installed Serial Ports and MIDI devices can be selected and assigned to the selected Console Interface:

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.

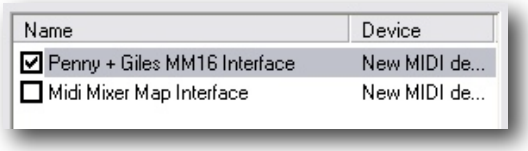
Right Click Menu



Right-clicking on a Console Interface opens a menu with five entries that match the functions of the five buttons just discussed.

Using Console Manager

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.

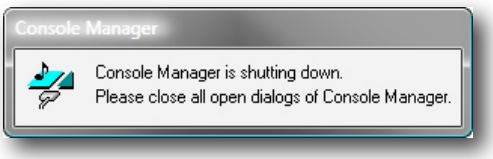
Saving the Console Manager configuration in the SSL Mixer software

Activating the option **Connect to Console Manager** in the menu: **Settings|Preferences|Mixer** initiates the basic communication betwen the SSL Mixer V& and the Console Manager Module.
Performing a **Save Settings** in menu: **Settings|Save Settings** in the SSL Mixer software 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 Mixer 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 which 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.

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

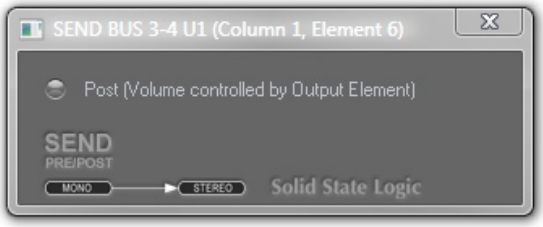
Several types of send elements are available: mono, stereo (with a balance control), mono to stereo (with a pan control), with or without Equal Power Panning.

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.

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 next element below inside the mixer strip.

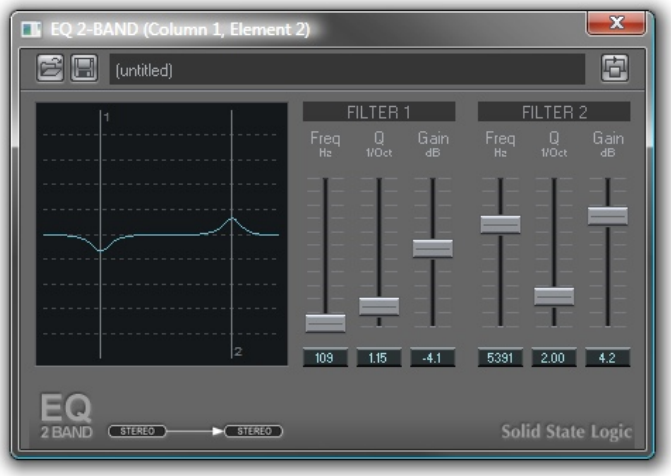
EQ 2-band

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 work 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.

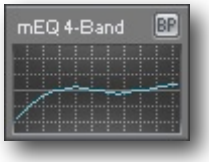


The EQ parameters can be adjusted in real-time by using the mouse buttons in the EQ parameters value boxes in the mixer strip or by using the scrollwheel. If you hold down both mouse buttons the EQ parameters can be changed more quickly, and the direction of the change is determined by the mouse button 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 resource 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 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 3 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.

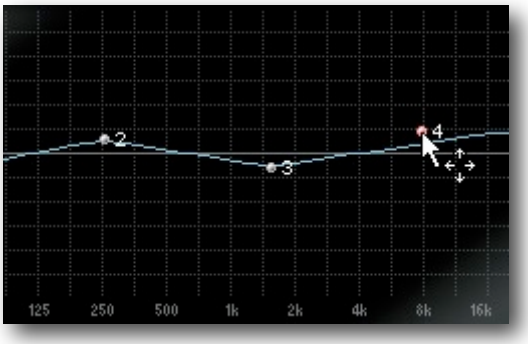
Left-clicking the "MUTE" buttons toggles between mute and unmute of the EQ band. The button is highlighted when the EQ band is muted.
Right-clicking a "MUTE" button "solos" the corresponding EQ band, meaning that the other EQ bands become inactive/muted. When an EQ band is "soloed", right-clicking the "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.

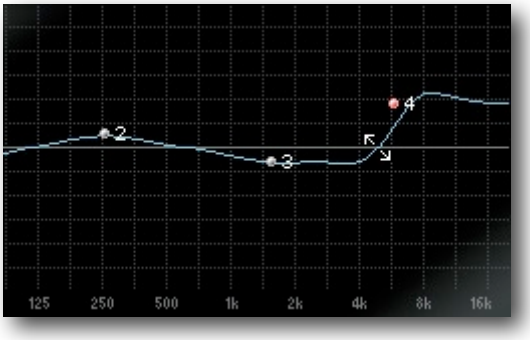
The EQ curve can be edited directly in the graph display:

- 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.

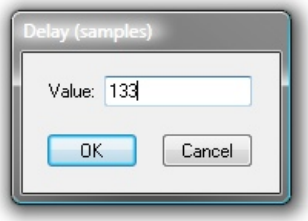
Sample delay line

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

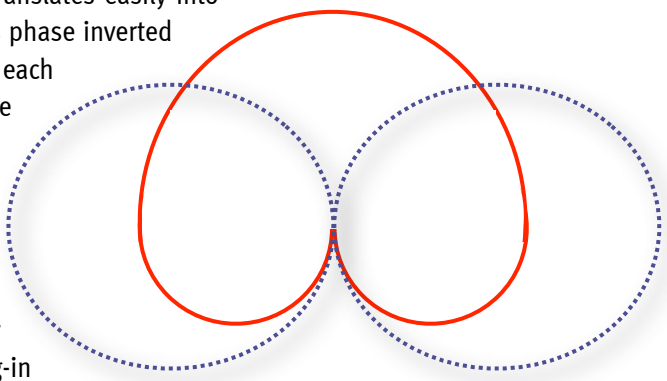
Mono and stereo peak meters can be inserted at any point inside a mixer strip.

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.



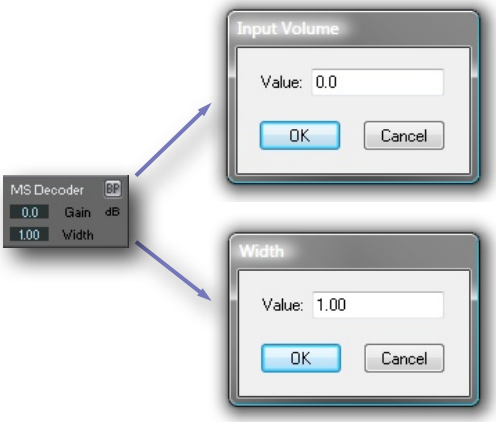
Cardioid microphone, mid signal
Figure of eight microphone, side signal

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.

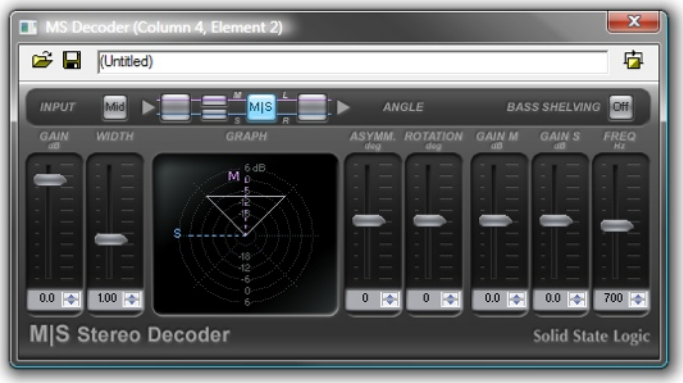
The **gain** and **width** parameters for the MS signal can be adjusted directly in the mixer strip using the mouse buttons or scroll wheel.

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's to open the main MS Decoder window.



Input section

Lock Mid button

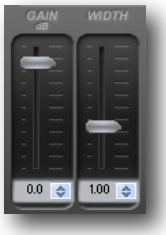
Lock Mid activated

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 button

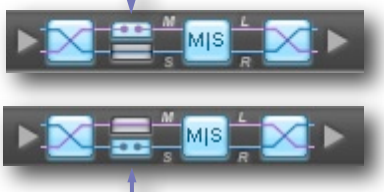
Swap Inputs
Swaps the input channels before the signals enters the MS matrix .

Swap Outputs button

Swap Outputs
Swaps the output channels post MS matrix .

Mid On/Off 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.



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.

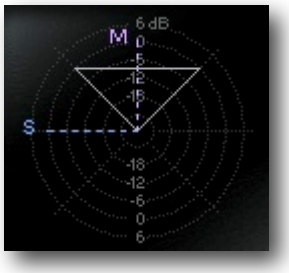
MS Graph

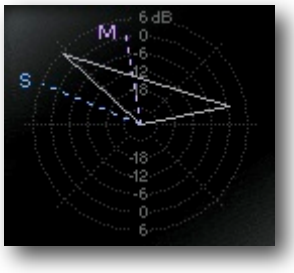
The graph display provides a visual representation of the perceived sound field, as defined and modified in real time by the parameter controls, and contributes to the intuitive operation of the MS Decoder.

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 (odB 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 Mixer software (you can find the password in the product's registration card and in a sticker on the MX4 box). Entering the password in 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 very useful collection of essential tools, bread and butter processes for everyday tasks.
More comprehensive, optional DSP effects are available from Solid State Logic and third party companies.

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).

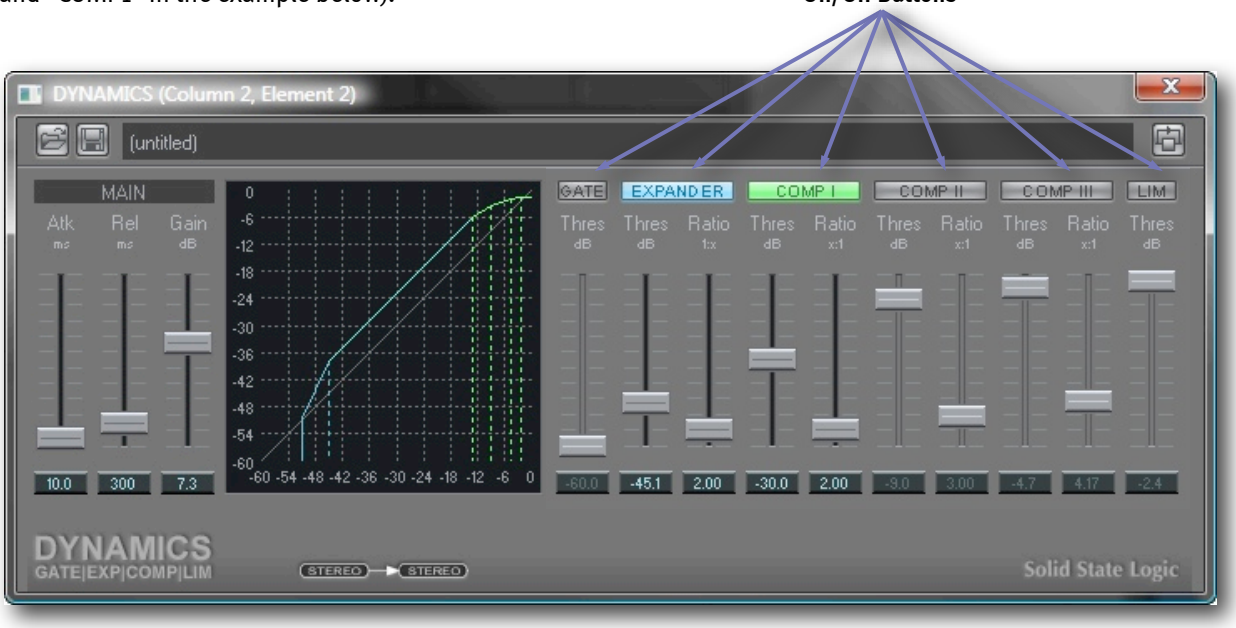
In the main section of the window, the "Mix" fader controls the amount of signal that is delayed and sent to the LFO, the "Delay" fader controls the length of the delay, and the "FdBk" (feedback) fader controls the proportion of the processed signal that is sent back the input.

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.

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 is on a decay.

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 oms 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 odB 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.

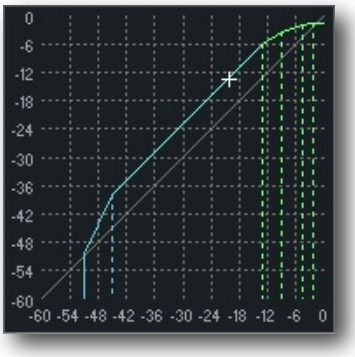
NOTE: The measurement of the input signal depends on the Attack and Release settings and higher values will slow down the animated gain indicator on the curve.

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 dwon. 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.

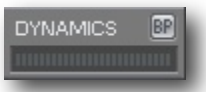
Moving any threshold fader with the right mouse button will move all other threshold faders relatively, while maintaining the dB relationship. This allows the whole curve to be shifted moved up or down in the graph.

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.

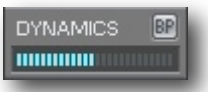


Gain Metering

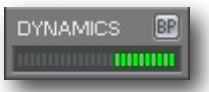
The amount of gain reduction is shown by the meter in the Dynamics mixer element. The meter calibration is 3dB per segment.



No Processing



Expansion



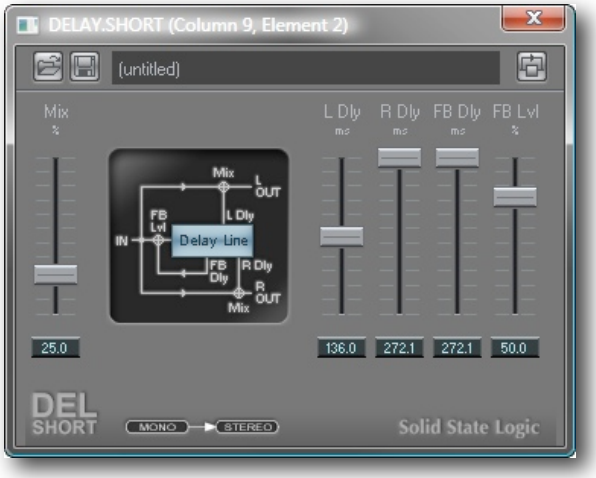
Compression

In the main "Dynamics" window, the level of the processed signal is shown in the curve display as a white "+" sign that moves in real time along the curve. This makes it easy to check whether the signal is operating at the correct point on the curve. The level of the input signal level is shown as a small vertical white line which moves along the horizontal axis .

Delay

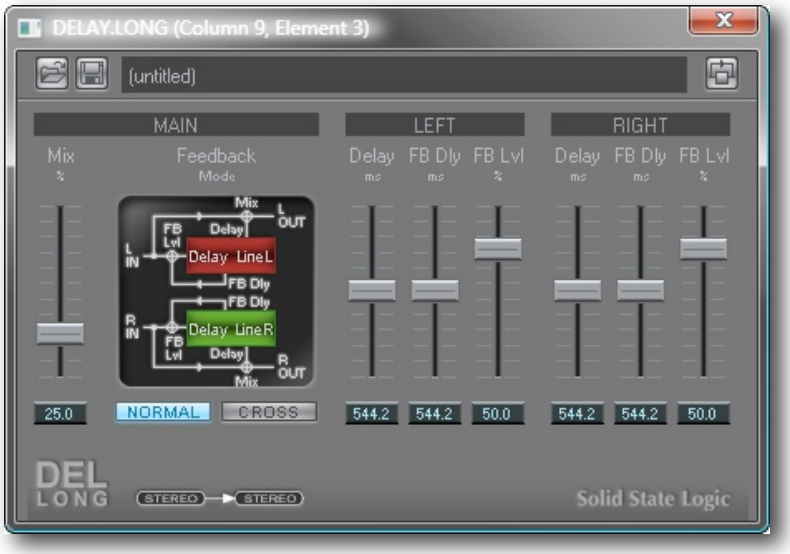
The Delay plug-in provides a 2-tap mono, 2-tap mono to stereo, or 2-tap stereo delay, with selectable cross-linking between left and right channels. Each is available as a Long, Medium or Short delay version, longer possible delays need more DSP Memory, therefore it is advisable to select the element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



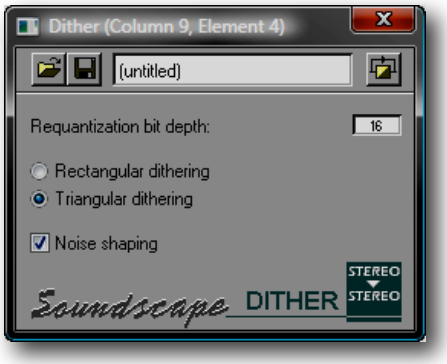
Clicking the "cross" or "normal" button for a stereo delay allow to 'cross couple' the feedback portion of left and right delay outputs for panning delay effects. In cross mode the feedback signal from the left delay line is fed into the input of the right delay line and vice versa.

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.



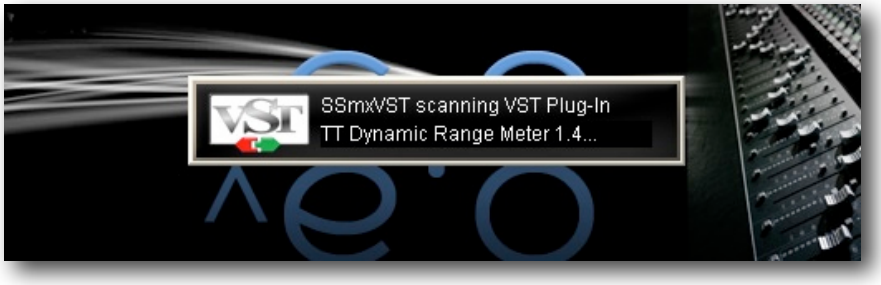
The "Requantization bit depth" can be changed in the value box using the mouse buttons in the usual way. Alternatively, double-clicking in the value box will open a a text entry dialogue.

Using VST Plug-Ins

VST effect plug-ins running on the host CPU can be inserted in the SSL 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 the Software during startup.
By default, the SSL Mixer 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 relevant .ini file (located in C:\Soundscape in a standard installation), for instance: "VSTLibPath=C:\Program Files\Steinberg\Vstplugins\".
Note that any subfolders of the designated folder will be scanned as well.

During scanning, a small window appears in front of the splashcreen where the plug-ins' names are displayed as they are loaded.

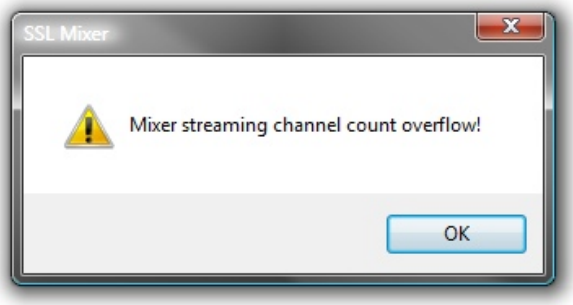


Dialogue boxes may also appear during that phase, for instance when a plug-in requires authorisation.

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.

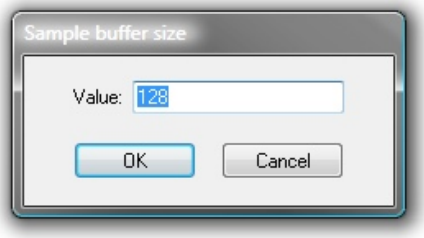
Since the VST plug-ins run on the CPU of the host PC, the audio must be routed at high speed between the SSL Mixer and the PC in order to use VST effects at low latencies.
This is done using the "streams" provided by the MX4 or Mixpander.
One stream is used per channel of audio, so that a mono VST plug-in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

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.



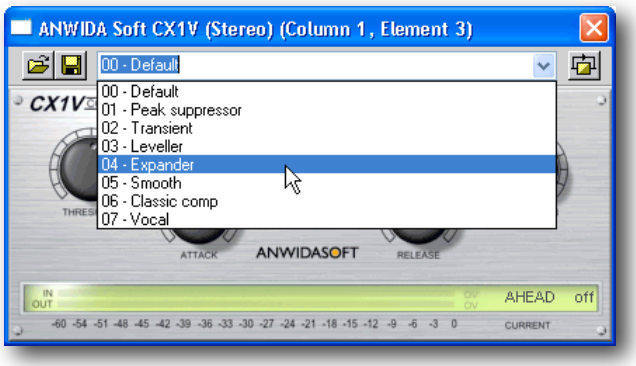
In this case the Mixer will also become inactive until the situation is resolved by deleting one or more VST plug-in(s), reassigning or muting one or more input, output or send element(s) or closing or re-routing a Windows application in order to release some streams.

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 build 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 unmuting mixer elements.

VSTi mixer elements

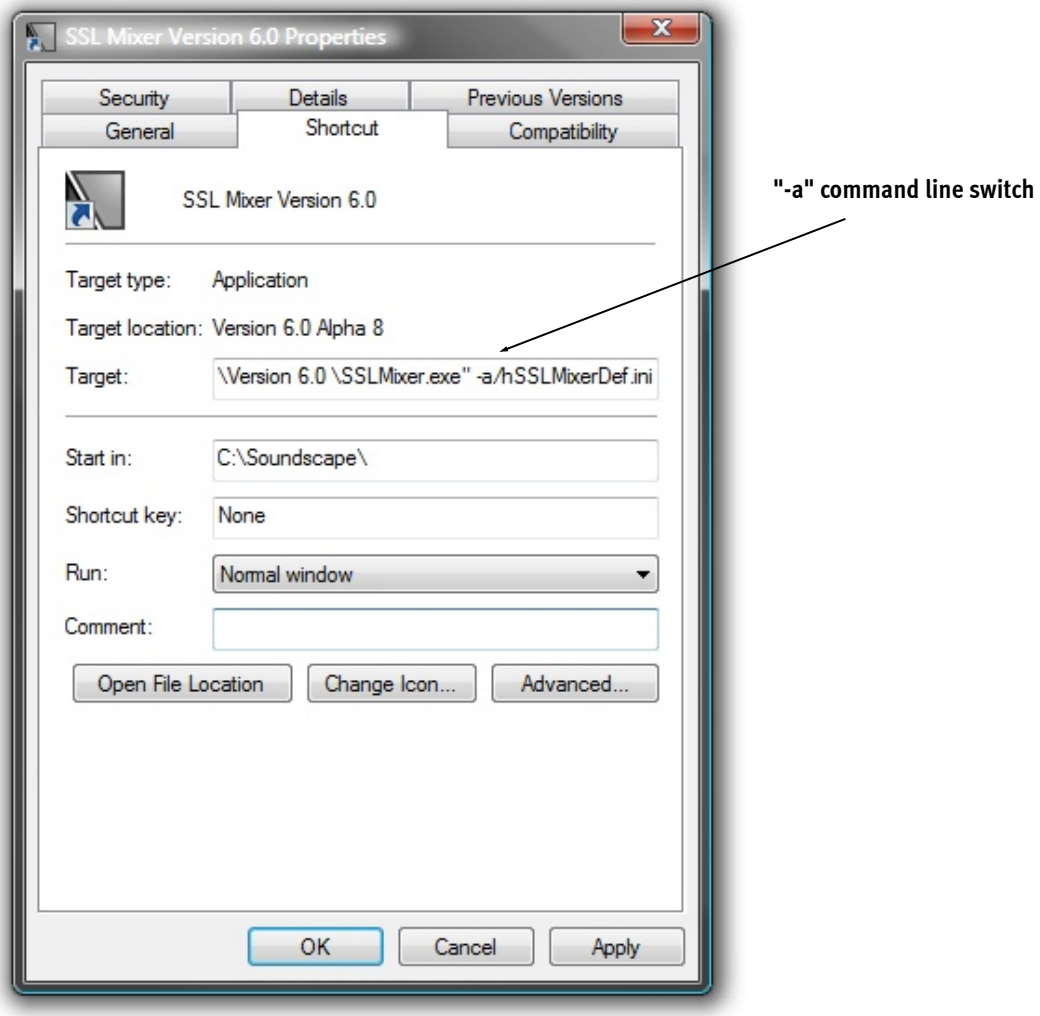
Everything mentioned in the previous section about VST plug-ins also applies to VSTi (VST instrument) plug-ins.

In addition VSTi plug-ins have "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 does would immediately close again.
At present the only tested examples of this are Lounge Lizard and Tassman from Applied Acoustics Systems but the same may occur with other plug-ins.
The command line switch solves the problem by initializing COM objects as "APPARTMENT_THREADED" instead of "MULTI_THREADED".
To use the "-a" command switch, insert "-a" just after "SSLMixer.exe" in the Target line of the SSL Mixer shortcut's properties.
In a standard installation the SSL Mixer's shortcut is located in C:\Documents and Settings\All Users\Start Menu\Programs \Startup.
Right-clicking the shortcut will open the Properties window. The edited line should read as follows:
"C:\Program Files\Soundscape\SSLMixer\SSLMixer.exe" -a /h SSLMixerDef.ini



NOTE: Using this command line switch may slow down the performance of the **SSL Console Manager application**.

Automatic Delay Compensation (ADC)

In a digital mixing system, any 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...:

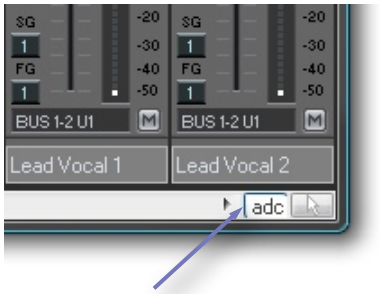
- **MADI1-2 U1,** used by the "Output (Stereo)" element in mixer column 1
- **MADI3-4 U1,** used by the "Output (Stereo)" element in mixer column 2

...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.

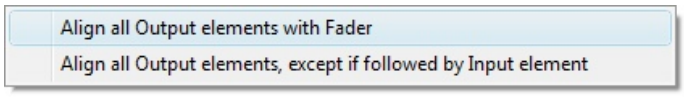
Automatic Delay Compensation toggle (ADC disabled)



Automatic Delay Compensation toggle (ADC enabled)

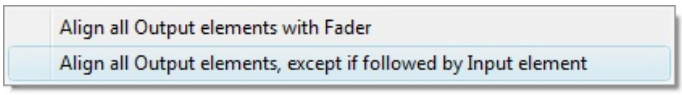
Alternatively, you can click the "Automatic Delay Compensation toggle" item in the **Edit menu** or use the **[Ctrl]+[Alt]+[D]** key command.

Align all Output elements with fader



In order to line up outputs that have isolated audio paths in the Mixer, the "Align all Output elements with fader" option in the menu:
Settings|Preferences|Mixer| If Automatic Delay Compensation is active can be used.
When this option is selected, all output elements that have a fader are lined up.
Outputs elements without a fader can be used for signals that should not be lined up (e.g., a send/return loop to an external effects processor).

Align all Output elements, except if followed by Input element



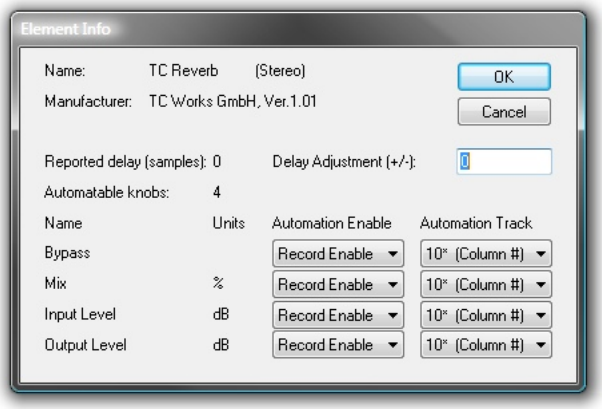
Enabling the "Align all Output elements, except if followed by Input element" option under menu:
Settings|Preferences|Mixer|If Automatic Delay Compensation is active will line up all the output elements, except those that are followed by an Input element.

Mixer element Delay Adjustment parameter

The ADC function relies on the delay values reported by each individual mixer element to calculate the compensating delays that must be applied. This value can be checked and edited by the user.

In Mixer Edit mode, clicking a mixer element with the Info tool...:

...will open the corresponding "Element Info" window, as shown on the right side 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 SSL Mixer .INI file (the path to the .INI file can be checked in the "Start in" field of the SSL Mixer shortcut's properties).

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: The Mixer 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:

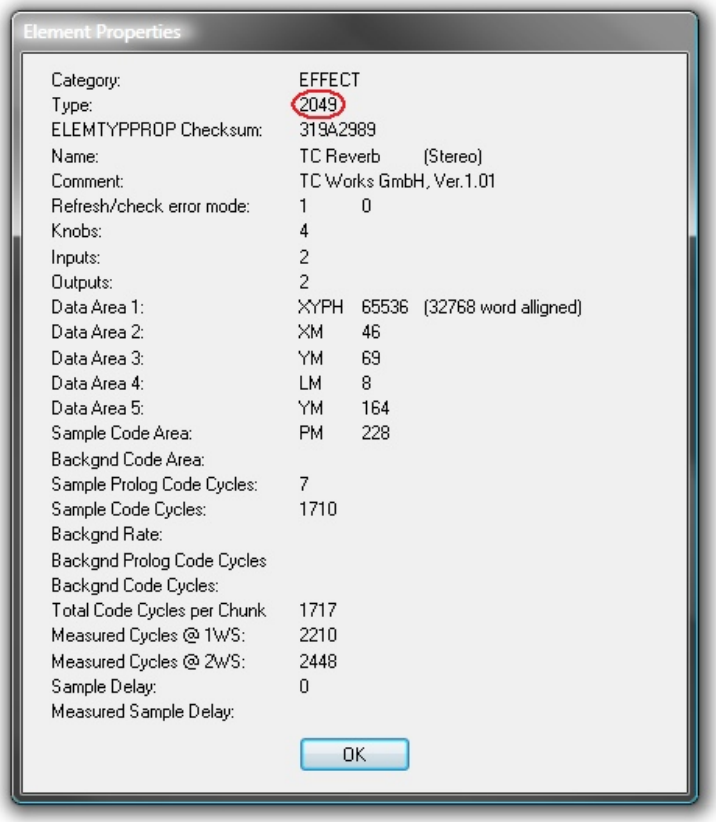
ElemType XXXXXXXX Delay=Y // Element's name

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 on the right side for the TC Dynamizer. The Element Type ID is shown on the second line (red circle).



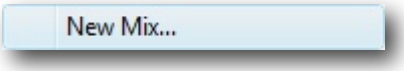
To continue with the same example, in order to specify a delay value of 45 for the TC Dynamizer stereo mixer element, the command line in the **[MixWnd]** section of the .INI file should read:

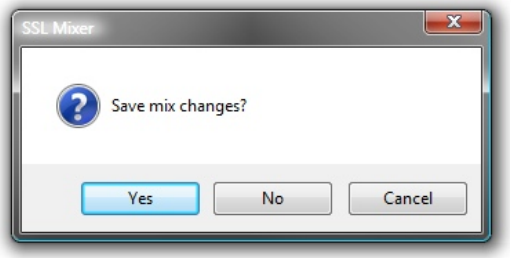
ElemType 2050 Delay=45 // TC Dynamizer (Stereo)

File Menu

New Mix

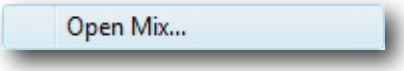
Clicking "New Mix..". will create a new, empty Mixer.
If a Mixer is already active and has been edited since the last save, the following dialog box will appear:



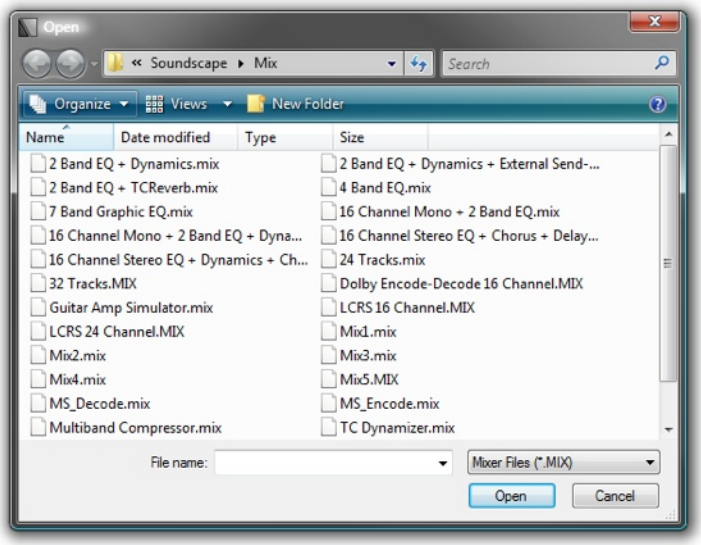


Open Mix

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

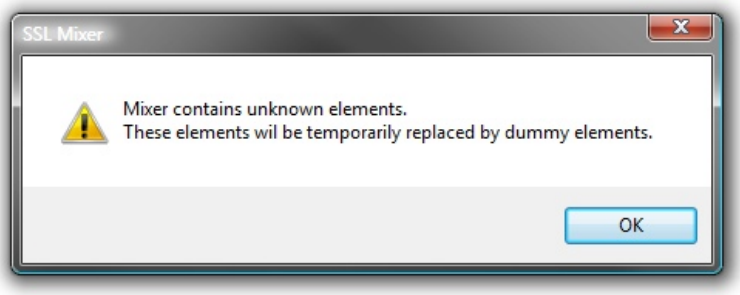


If a Mixer is already active when you open a new Mixer, the same dialogue box above will appear asking you to first save your current Mixer.

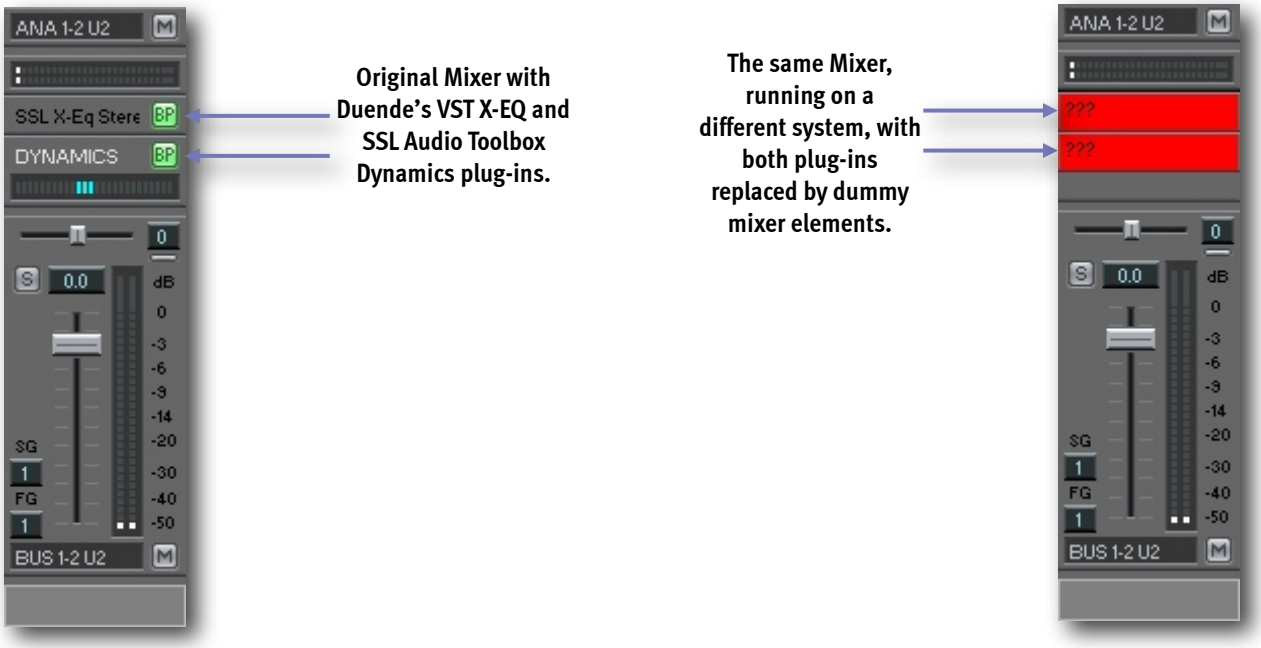


Opening a Mixer: special cases
Loading a Mixer on the system where it was created is straightforward.
If you are attempting to load MIXes on a different SSL system containing different hardware or software configurations (e.g., different number of hardware units or plug-ins that are not available).

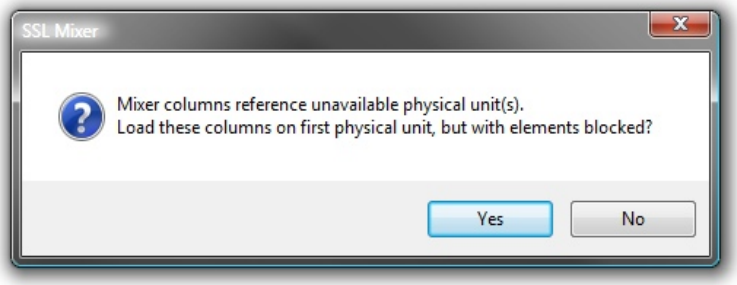
Whenever possible, if the Mixer you are opening includes one or more mixer elements which are not available on the



The unknown mixer elements will be replaced by dummy mixer elements, giving you a chance to locate and correct the problem (s) easily. In the example below, the Mixer was created on a system where the SSL Duende's VST X-EQ and SSL Audio Toolbox Dynamics plug-in were available.
The same Mixer was loaded onto a system where these plug-ins are not available shows the missing plug-ins as red dummy elements:

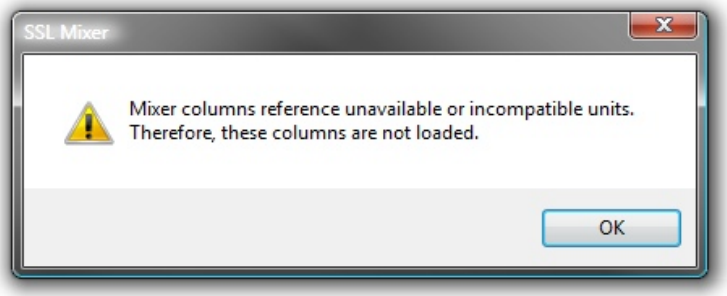


If the Mixer you are opening contains columns which refer to SSL physical units (i.e., hardware units) that are not available, the following dialogue box will let you decide to if these columns should be loaded with their elements blocked (i.e., muted) or skipped:



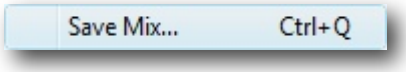
Clicking **Yes** will load these mixer columns. The Mute tool can be used to unmute them.
These columns will be assigned to the highest numbered unit within the system. This may cause mixer elements to become muted in other mixer columns.

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



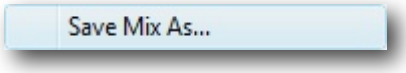
Save Mix [Ctrl]+[Q]

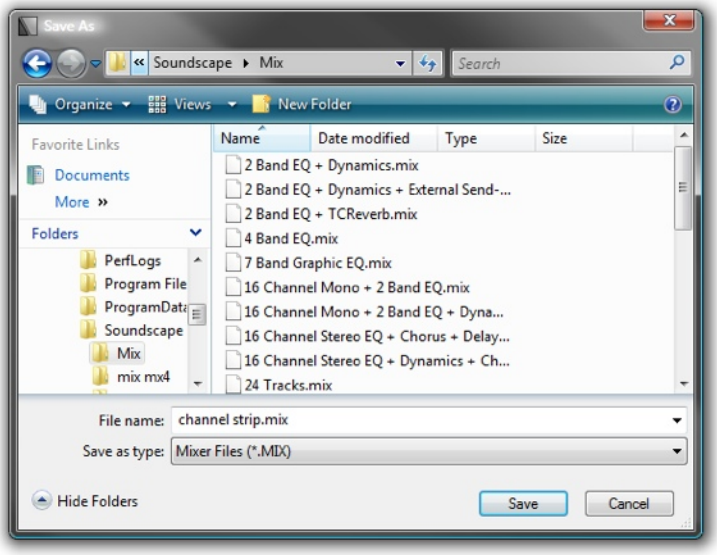
"Save Mix" saves or updates the active Mixer. If the active Mixer has not been saved previously, a standard Windows "Save As" browser dialogue opens.



Save As

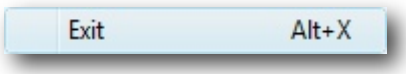
Clicking "Save As" a standard Windows "Save As" browser dialogue opens, the Mixer needs to be named or renamed and you can browse your local and network drives, to select the destination path.

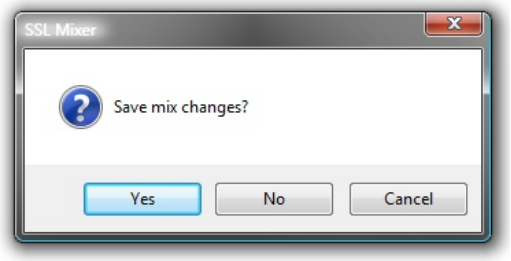




Exit

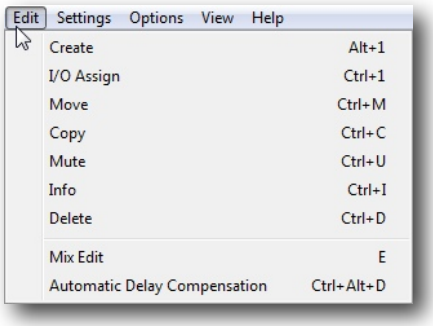
"Exit" closes the program. If the currently active Mixer has been modified since the last save operation, the following dialog box will appear:



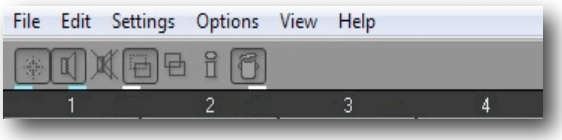


Edit Menu

The first section of the Edit menu lists the Mixer editing tools, and clicking on any item in this section assigns the corresponding tool to the left mouse button.



You can also access these tools directly from the Toolbar:

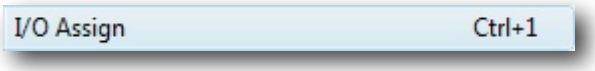


NOTE: The Edit Tools only work, when the Mixer is in **Mix Edit mode,** (Edit|Mix Edit or by pressing [E] on the keyboard).

Create [Alt]+[1]

Clicking with the Create tool in an empty "mixer slot" will create a "mixer column". Clicking in the user definable space in a "mixer column" will open the list of "mixer elements" which can be inserted. This tool is used as the basic building procedure for the SSL Mixer.

I/O Assign [Ctrl]+[1]

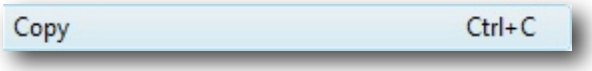


This tool selects the input or output routing of all mixer element that have an I/O label. The I/O assign tool also handles the DSP assignment by clicking in an empty column area.

Move [Ctrl]+[M]

The Move tool can be used to drag a mixer strip to a different column number or move an element inside a strip.

Copy [Ctrl]+[C]



Copy is similar to the Move tool, except it adds a new element or column at the copy destination.

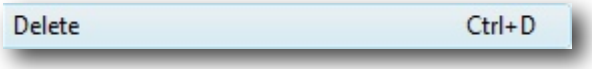
DSP Mute [Ctrl]+[U]

The DSP Mute tool is used to mute or unmute mixer columns or mixer elements.

Info [Ctrl]+[I]

The Info tool opens the Element Property window, which contains relevant settings for automation control, ADC settings and general Information about the Plug-In.

Delete [Ctrl]+[D]



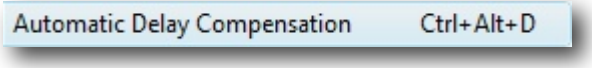
The Delete tool is used to delete mixer columns or mixer elements.

Mix Edit [E]

Clicking this this entry or pressing [E] on the computer keyboard switches between Mixer Edit and Mixer Control modes.

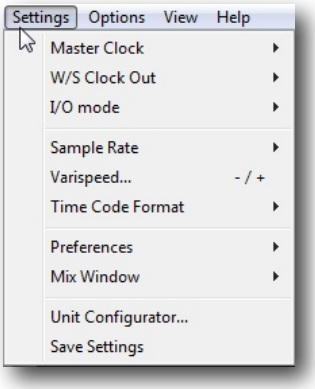
Automatic Delay Compensation [Ctrl]+[Alt]+[D]

Clicking this item or pressing [Ctrl]+[Alt]+[D] on the computer keyboard enables/disables (toggle) Automatic Delay Compensation.



Settings Menu

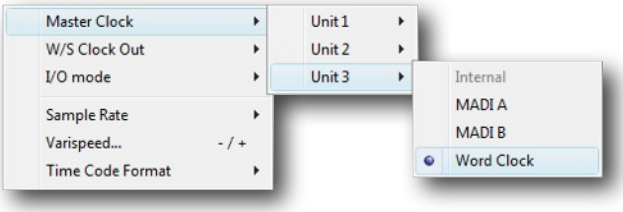
Master Clock



Clicking on "Master Clock" opens a submenu for each installed SSL device (displayed as Unit 1, Unit 2...).

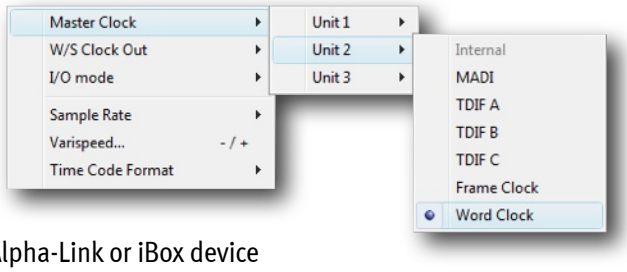
Available clock sources vary depending on device type and interface they are connected to (Mixpander).

Master clock settings for a 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, that 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 Houseclock Generator, that has multiple direct Word Clock Outputs and syncs the whole facility to this stable clock.

Master clock settings for a Mixpander and SSL Alpha-Link/ Soundscape iBox units

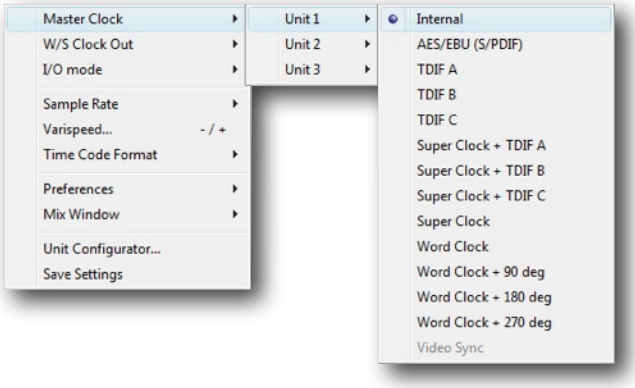


Since Mixpander systems can contain various models of the Alpha-Link/iBox Product Range with several digital I/O options, the available Master Clock options vary (and only depend on the I/O configuration of the connected converter box).

- **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. In order to achieve the double amount of information 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 Houseclock 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 with Soundscape 32 unit

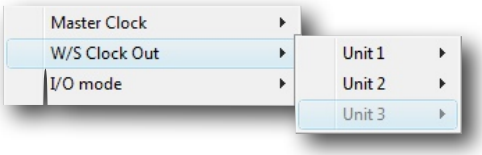


- **Internal**
  When this option is selected, the Sample Rate is set by the Soundscape unit's on board Master Clock Generator. This should be used in conjunction with analogue equipment only.
- **AES/EBU (S/PDIF)**
  When this option is selected the Soundscape 32 is slaved to the WordClock signal received 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 system.

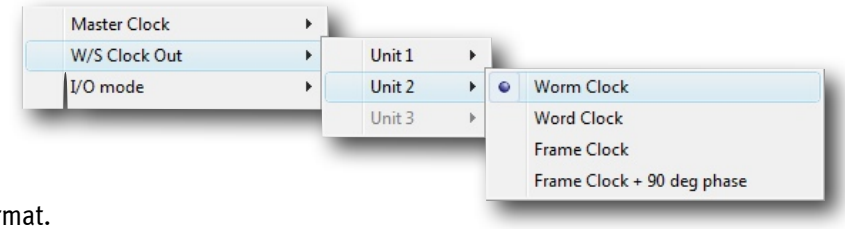
W/S Clock Out settings for MX4



The MX4 has one BNC Word Clock connector that 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 (Unit 3 on the right).

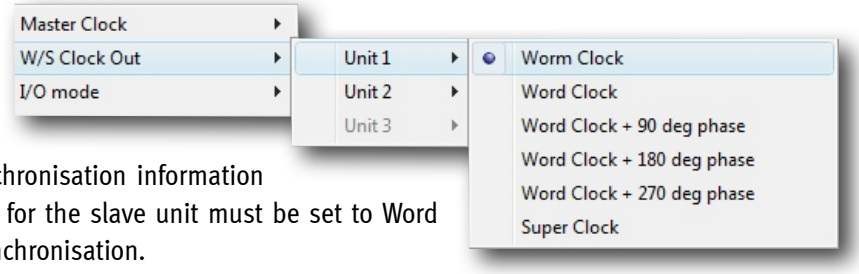
W/S Clock Out settings for Mixpander and Alpha-Link/iBox units

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.



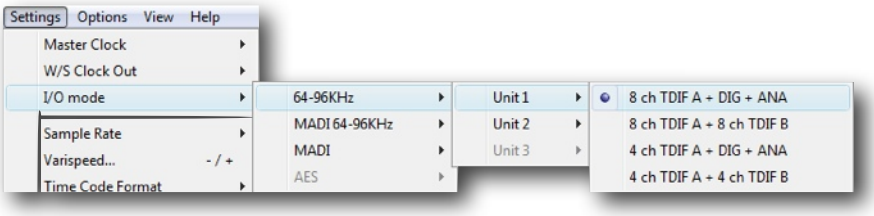
- **Worm Clock**
  SSL's proprietary extension of the Word Clock Format.
  Worm Clock is a modified Word Clock signal that includes sample accurate start/stop synchronisation information for several units. The Master Clock parameter for the slave unit must be set to Word Clock, which also enables Worm 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 that Word Clock is varying in frequency (in Chase Slaving conditions), please use Super Clock instead.

When the mouse pointer is on "I/O mode", several submenus open that allow hardware specific settings to be made for operation at high Sample Rates and MADI Ch Modes.



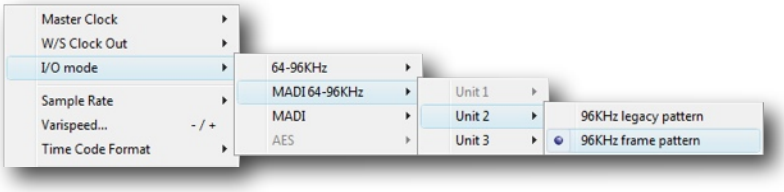
Again, the available options are determined by the SSL Hardware you are using.

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.

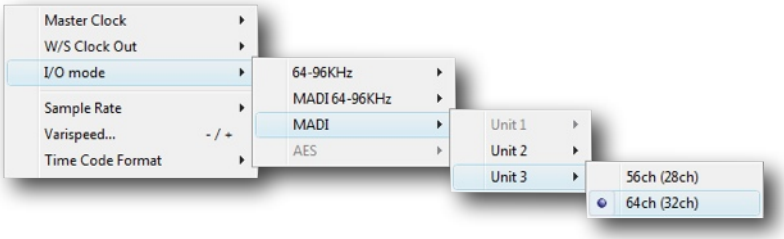
MADI 64-96kHz

In this submenu the MADI Frame Pattern is selected between **96k Legacy Pattern** 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).

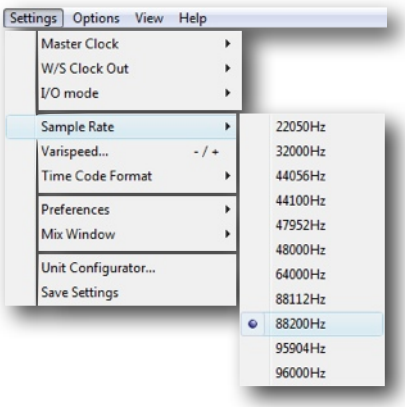


Later the **64 CH MADI Mode** was introduced as an AES standard, which is always using the full MADI data bandwidth, hence doesn‘t allow a higher sample rate.

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.

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.

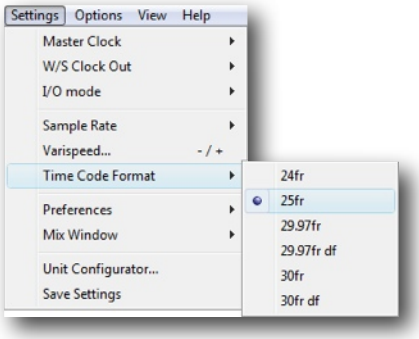
Internal Sample Rates:

- **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)

Varispeed

The Varispeed setting provides a +/-10% variation of the internal Sample Rate. Clicking "Varispeed" opens the Varispeed dialogue box where percentage (- or +) can be entered.)

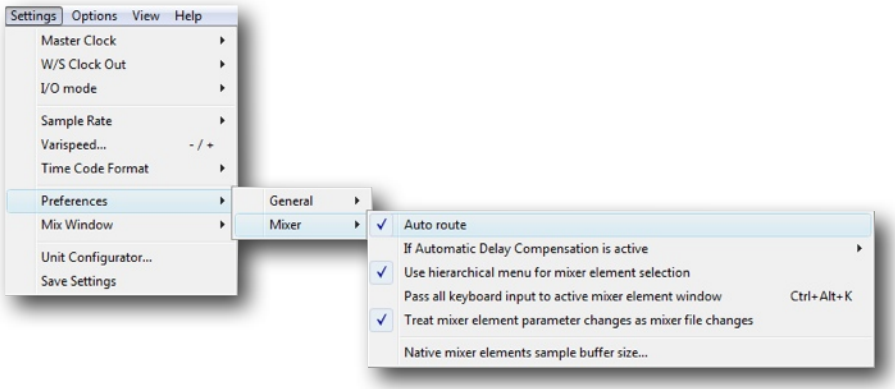
Frame Rate



Frame Rate selects any of the commonly used 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. The SSL Mixer V6 only uses this setting when a Video Sync capable SSL Device is part of the System (like Soundscape 32/Sync Card/PCi Host/Mixpander).

Preferences

This entry is divided into two sub menus, **General** and **Mixer**.

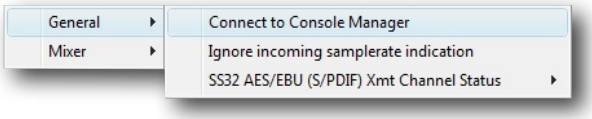


Most entries inside the Preferences Menu are options, that can be activated (ticked) or deactivated. Only a few items (arrow or dot to the right) open further submenus or dialogue boxes.

General sub menu

Connect to Console Manager

Please check the Console Manager Installation in Chapter 5.

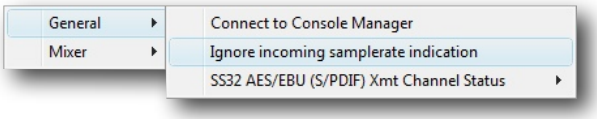


Activating this entry connects the SSL Mixer V6 to the optional Console Manager application, which can be used connect the SSL Mixer Software with a huge variety of hardware control surface.

If the Console Manager Module is not already active it will be launched automatically.

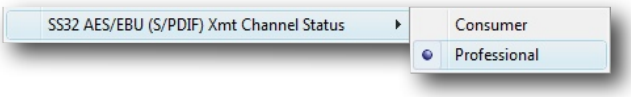
Please read the Section **Controlling the Mixer with a Hardware Control Surface** for further details on how to set up and use the Console Manager Module.

Ignore Incoming Sample Rate Indication



When this option is selected, the incoming Sample Rate indication is ignored, and the currently selected Sample Rate is maintained .
This is useful if the Master Clock source is transmitting an incorrect Sample Rate indication.

AES/EBU (S/PDIF) Xmt Channel Status

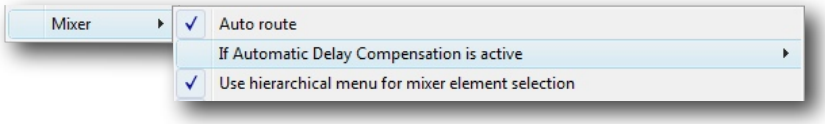


You can set AES/EBU or S/PDIF to communicate in a "Consumer" or "Professional" format for the digital audio data transmitted from a Soundscape 32's AES/EBU (S/PDIF) outputs.
For example, some professional DAT recorders will not record from their digital inputs if they receive data in the "Consumer" format.
Soundscape 32 adjusts to either format automatically, therefore there is no equivalent setting for the AES/EBU (S/PDIF) digital input.

Auto Route Mixer

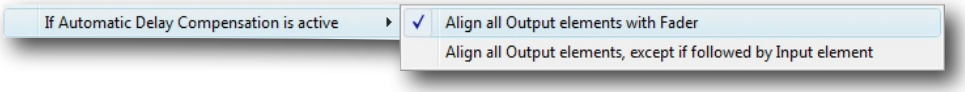
Activating this option enables the Mixer Auto Router to automatically assign available DSP Resources to inserted Columns or elements. Please check the Section **DSP resources and DSP Auto Routing** in **Chapter 8** for further details.

If Automatic Delay Compensation is active



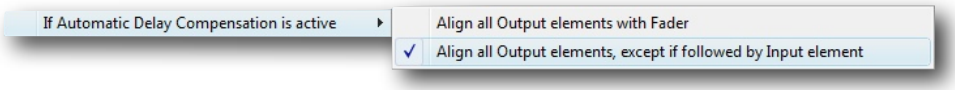
ADC offers several options to allign or not alter certain outputs in time.
Please check the Section **Automatic Delay Compensation** in **Chapter 8** for more details.

Align all output elements with fader



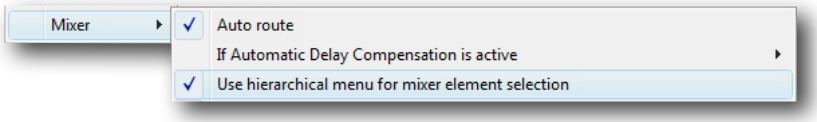
In order to line up outputs that have isolated audio paths in the Mixer, the "Align all Output elements with fader" option can be used.
When this option is selected, all output elements that have a fader before the Outputs are time aligned. Outputs elements without a fader can be used for signals that should not be aligned/delayed (e.g., a send/return loop to an external effects processor, Artist Monitor etc.).

Align all output elements with fader except if followed by input element

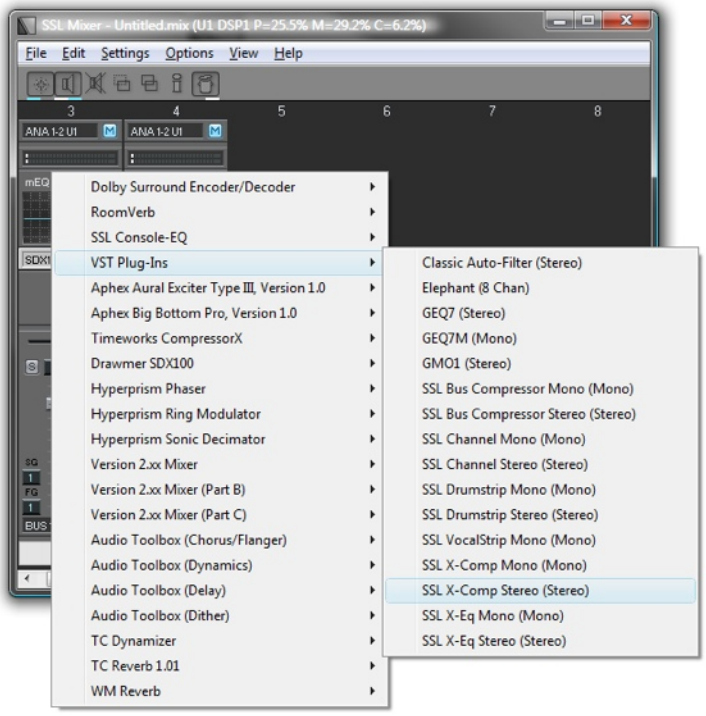


This option will line up all output elements, regardless if they have a fader or not, but it will exclude Outputs that are followed by an Input element (ie. a send/return to external hardware).

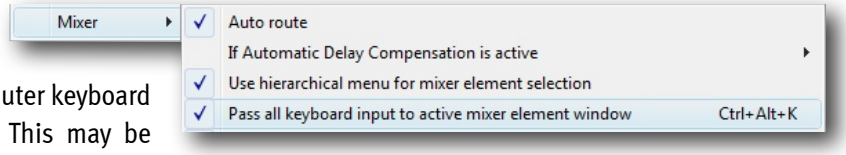
Use hierarchical menu for mixer element selection



If activated, this option will sort all entries hierarchically that appear in the **create element list,** while inserting a mixer element into a mixer column.

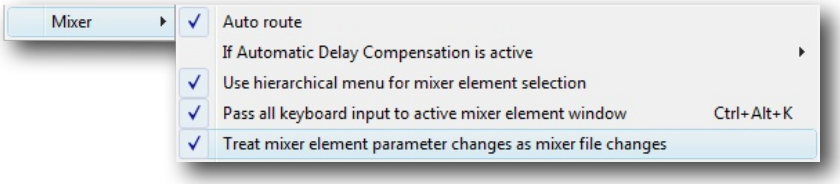


Pass all keyboard input
to active mixer element window



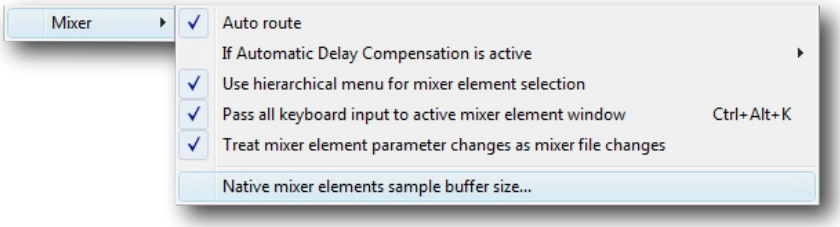
When this option is activated, input from the computer keyboard is passed to the active mixer element window. This may be necessary in order to authorise or use certain shortcuts in a VST Plug-In. Please note, that some global SSL Mixer V6 keyboard commands may not work anymore, so it's probably best to use this option only if required.

Treat mixer element parameter changes as mixer file changes



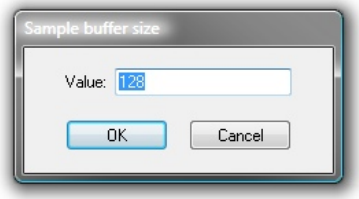
When this option is enabled, a change of mixer element parameter will enable the Save Mix function in the File menu. If it is disabled, only changes to the structure or routing of the Mixer enable the **Save Mix** function.

Native mixer elements Sample Buffer Size

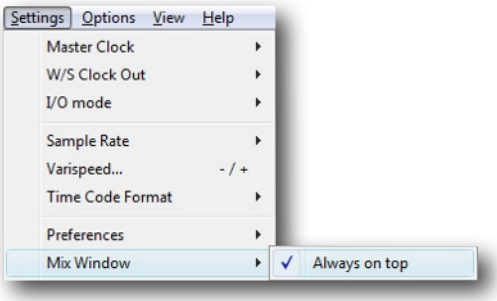


This entry opens a dialogue box to enter the Sample Buffer Size for native mixer elements (VST/VSTi Plug-Ins).

The default buffer size for native mixer elements is set to 128 samples. Increase this number if your PC is struggling to cope with the processing demands. Decrease this value, if you need lower latencies (processing delays) while Monitoring through a native mixer element. Multiples of 64 up to 8192 samples can be used.
If an invalid value is entered, the closest valid value will be used.

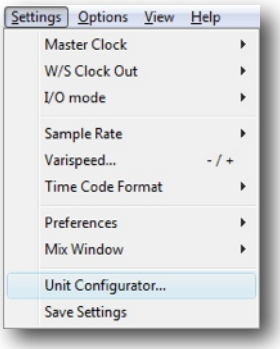


Mix Window



If "Always on top" is activated, the SSL Mixer window will always float on top of any other window, even if that other window is active and does not have a similar option activated.
When working with a single monitor, this option can be quite confusing, since you have to minimise the Mixer V6 application in order to fully access other applications that are only partly covered by the mixer window.

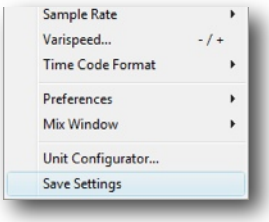
Unit Configurator



This entry opens the Unit Configurator window, where you can enable, disable or re number unit id's of all Mixer compatible SSL devices installed in your system.

The Unit Configurator is described in detail in the section **Configuring a multiple unit system with the SSL Mixer V6** in **Chapter 6.**

Save Settings



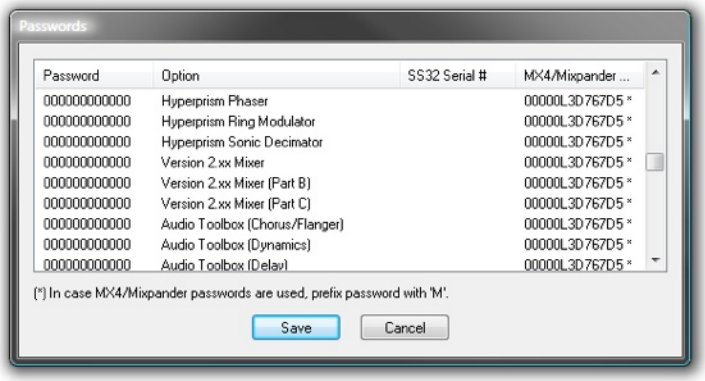
Any changes you have made in the Settings Menu (and all sub menus) are only saved as default, when you click this option.
Without executing **Save Settings**, the SSL Mixer V6 will on next startup revert to the previous settings. Some Parameters, such as the current window size and screen position, are saved as part of the mixer file.

Options Menu

Clicking on Options opens the Password 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 Mixer 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 be copied to the other list entries.

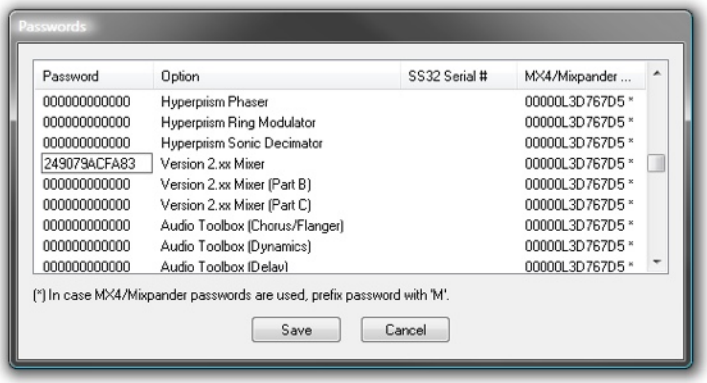
The Mixer Password for a new SSL hardware unit can be found on a sticker that is placed directly on the board or on the product packaging and registration card. It can also be obtained from our website if you have registered the unit.



Initially, all the passwords are set to "oooooooooooo", as shown above.
When entering a plug-in password generated for an MX4 card or Mixpander, a leading "M" needs to be entered. If several cards are present in the system, make sure that the cards serial number in the right column matches with your password entry.

To enter a password simply double click on the password text field.

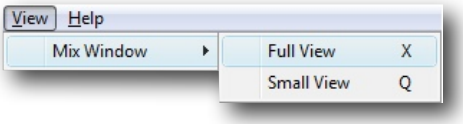
When all the required passwords have been entered, click the "Save" button to store the settings.



NOTE: 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 the SSL Mixer. The password should only be entered for the unit it was generated for, the other unit(s) should be left at the default "oooooooooooo". Only then Password Sharing will allow to work with this module on all Units.

View Menu

Inside the View Menu you can select one of two different Mixer Views, Full View (wide Channel Strips) and Small View (Narrow Channel Strips).



Small Column Size [Q]
Click on **Small View** (or press **[Q]** on the computer keyboard) selects a **narrow view mode** for the SSL Mixer window.

More mixer columns can be displayed.
(17 columns at 1024 x 768 resolution)

but some controls, ie. the **solo group** and **fader/mute group assignment boxes,** are **hidden**.

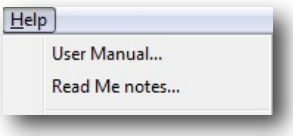
Full Column Size [X]
Click on **Full View** (or press **[X]** on the computer keyboard) selects a **wide view mode** for the SSL Mixer window.

Less mixer columns can be displayed
(11 columns at 1024 x 768 resolution)

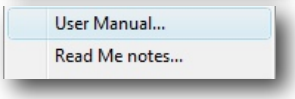
but **all** controls are exposed.

Help Menu

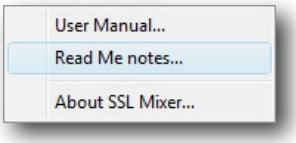
The Help menu allows direct access to this User Guide, the Readme Notes including last minute changes, and the About SSL Mixer Window, that displays additional SSL System Information.



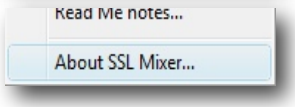
User Manual
Clicking on User Manual opens this reference Guide in a PDF Format. Please make sure that you have a PDF Reader installed on your System, otherwise Windows may display an Error message, that it could not open this file Type.



Read Me notes
The Readme Notes contain the latest changes in software revision.
It may contain important last minute changes, that did not yet make it into the User Manual.



About SSL Mixer
Opens the Window below.



Apart from the current Software Revision all installed SSL hardware 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.

Click **OK** to close the window.

8. Support

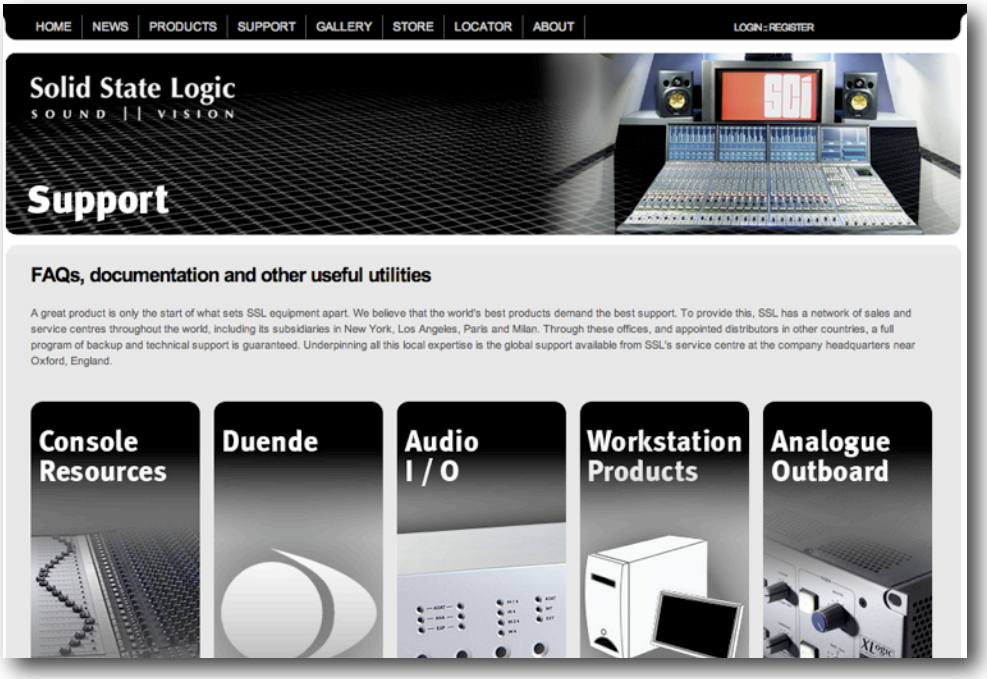
Support, FAQs and Online Help Centre

To access the latest support information on SSL 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 make sure all information is accurate.
All information is available to you 24/7/365.

If you can't find your 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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E&OE