Stereo Toolbox Audio Processor
User Manual

Safe Sound Audio

Stereo Toolbox Features

500 Series
Fully compatible with the API™ 500 Series modular format which allows a number of audio modules to be plugged into rack frames such as the API™ 6-slot Lunchbox and 10-slot 500VPR rack frame.

Quality precision stereo compression
- Compress in LEFT/RIGHT or MID/SIDE
- Side-chain filtering for that big bass sound
- Glue that 2-buss into the mix with confidence – every time

Three modes of operation
- **L/R link** : audio is processed as left/right with the compressor working in stereo linked mode. Left channel controls both L and R compression and output levels
- **L/R** : audio is processed as left/right with separate compressor and gain controls for left and right
- **M/S** : audio is processed as mid/side with separate compressor and gain controls for mid and side. Audio is converted back into left and right before leaving the unit

Stereo width control and MID/SIDE processing offer a whole new palette of tonal shaping
- Bring dull mixes to life
- Compress centre image whilst retaining full stereo dynamics
- Precise control of stereo effects elements within the mix

Easy to setup and monitor
- Precise 41 position metal control knobs
- Recall your favourite control settings
- Front access fine calibration of audio levels
- LED monitoring of levels and gain reduction

Cinemag steel-core output transformers
MAGNA HOT +28dBu output levels available, selectable to super transparent or hi-color tones

Contents

500 Series Compatibility

The Stereo Toolbox Audio Processor is intended for use within an API™ 500 Series compatible rack frame from which is takes its power and audio connections. It cannot function as a stand-a-lone unit. Information about the API™ 500 Series rack frames can be found at www.apiaudio.com.

API is a registered trademark of Automated Processes Inc.

Installation

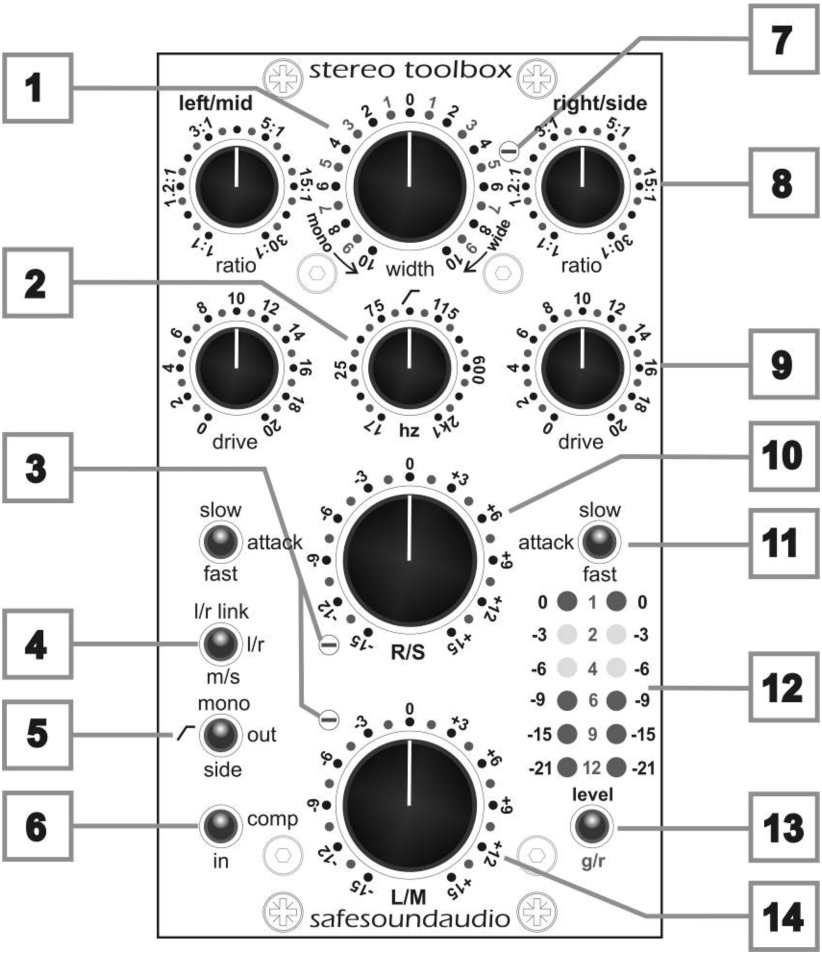
The Stereo Toolbox is supplied in a foam lined shipping carton and should arrive to you in perfect condition, but it is always worth checking the unit for any signs of physical damage before use. Should you have a concern that the unit appears damaged then contact your dealer.

The unit is quite heavy due to the inclusion of two high quality Cinemag output transformers, so take care when removing the unit from its foam lined carton and when handling the unit.

Ensure that your 500 Series compatible rack frame is powered off before inserting the Stereo Toolbox module. Select a spare double slot and remove the module mounting screws at that slot location. Slide the Stereo Toolbox module into the slot ensuring that the gold plated edge connectors on the module align with the corresponding connectors in the rack frame. Now gently push home the module until the front panel is flush with the rack frame. Use the four screws supplied with the rack frame to secure the module. Never use excessive force when inserting a module into the frame, so if you are struggling to get the module to insert, recheck the alignment and try again. Avoid touching the gold plated edge connector as you could tarnish the connects and reduce the reliability of the audio connection between the module and the rack frame.

Turn on the rack frame power, connect audio and check through the functionality of the Stereo Toolbox module according to the guidance in this manual.

Detailed operational guide



Audio Connections

For best performance, the Stereo Toolbox should be connected to a balanced line level audio system, but both inputs and outputs can be connected to line level unbalanced sources or destinations with no loss of level.

The unit is not designed to directly accept either microphone or unbalanced instrument inputs.

Although the unit may be switched to process signals internally as mid and side, the unit is designed to input from left and right audio and to output audio as left and right, when dealing with stereo audio.

Line level inputs should be connected to the female XLR input connector for the corresponding channel on the rear of the rack frame.

Line level outputs should be connected to the male XLR output connector for the corresponding channel on the rear of the rack frame.

The standard XLR pin allocations are used, that is

Pin 1 = screen
Pin 2 = signal +ve (also called ‘in-phase’ or ‘hot’)
Pin 3 = signal –ve (also called anti-phase’ or ‘cold’)

It is acceptable to connect pin 3 to screen when connecting to or from unbalanced audio sources.

Modes of Operation

The unit can be set to operate in three modes of operation according to how the three way toggle switch **(4)** is set, as follows;

Toggle up : L/R Link
Audio is processed as left/right with the compressor working in stereo linked mode. Left channel controls both L and R compression including output level control.

Toggle mid position : L/R
Audio is processed as left/right with separate compressor and gain controls for left and right audio channels.

Toggle down : M/S
Audio is still accepted as left/right but processed internally as mid and side with separate compressor and gain controls for mid and side audio channels. Level metering is switched to show internal mid and side audio levels. Audio outputs are converted back into left and right before leaving the unit

It is important to remember that the three modes of operation will affect how the audio signals are processed and how controls will operate.

High Pass Filter

The high pass filter is sited in the audio signal path immediately after the input stage but before the compressor. It is sweepable from 17Hz and 2.1kHz with a slope of 12dB/octave.

The filter can be set to operate in three modes of operation according to how the three way toggle switch **(5)** is set, as follows;

Toggle up : Mono
The filter is placed in the **mono control sidechain** of the compressor and is used to make the compressor less sensitive to bass

frequencies. This can reduce compressor pumping and retains more of the ‘big bass’ sound which is popular in music production.

Switching the filter to **mono** is not allowed when the Stereo Toolbox is set to the L/R (unlinked) mode of operation. In this case the filter will be switched off.

Toggle mid position : Out
The filter is switched out of circuit.

Toggle down : Side
The filter is placed in the main SIDE audio path and is used to remove unwanted low frequencies from the stereo content of the audio. This ensures that sources with heavy low frequency content such as kick drums are locked solidly to the centre of the stereo soundfield.

Switching the filter into **side mode** is only allowed when the Stereo Toolbox is set to the M/S mode of operation. In other modes of operation (when the filter is toggled down) the filter will be switched off.

Stereo Width Control

The stereo width control **(1)** is available in all modes of operation.

For normal width stereo set the rotary control to the 12 o’clock position (labelled ‘0’ on the front panel).

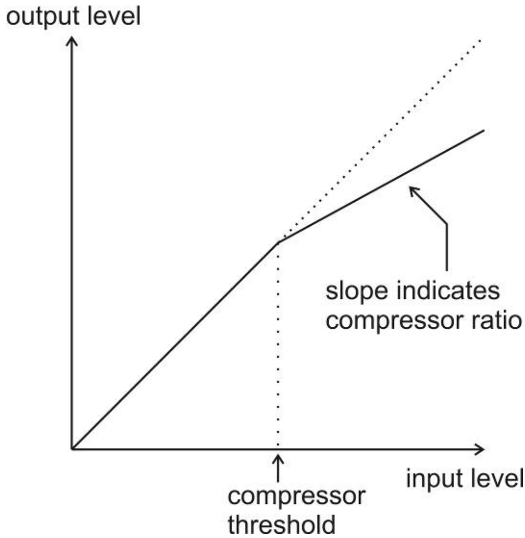
Turn the rotary control **anti-clockwise** to reduce the stereo content of the signal. When turned fully anti-clockwise the signal will be reduced to full mono.

Turn the rotary control **clockwise** to increase the stereo content of the signal. When turned fully clockwise the signal will have a much increased stereo width.

Examples of how to use the stereo width control during mixdown are described later in the user manual.

Compressor Controls

Compressors progressively reduce the gain of an audio signal as its input level rises above the compressor threshold as shown in the diagram below.



The ***compressor threshold*** is the level point in dB above which the audio gain will be reduced. On the Stereo Toolbox this can be varied from 0 to 20 using the **DRIVE** control **(9)**. Turning the control clockwise (from 0 towards 20) lowers the threshold and increases the amount of compression.

The ***ratio control*** **(8)** varies the degree of gain reduction which is applied, from 1:1 (no effect) to 30:1 which will make the compressor act like a limiter.

The ***attack time*** is how quickly the compressor will react to audio which rises above the threshold point. In the Stereo Toolbox the attack time can switched from 10ms (slow) to 1ms (fast) using the **ATTACK** toggle switch **(11)**. The setting of the attack time usually has a large impact on the compression effect.

The ***release time*** is how quickly the compression effect is removed when the audio falls back below the threshold. The Stereo Toolbox release time is programmed to follow the dynamics of the falling audio level. This is an important aspect of the unit's compressor design and offers a very natural sounding form of compression.

The amount of gain reduction (due to the compressor) is shown in the 6 segment LED meters **(12)** when the toggle switch **(13)** is selected downwards to **g/r**.

Bring the compressor into circuit by selecting toggle switch **(6)** downwards.

Because compression is a gain reduction tool, it will tend to lower the maximum audio level through the audio chain. Make-up gain is used to replace this 'lost' level and in the case of the Stereo Toolbox this is done using the rotary controls **(10)** and **(14)**. These controls have a range of ±15dB.

These gain make-up (also called output level) controls have an option to be in circuit at all times or can be set to operate only when the compressor is switched in. See the **internal module options** section in this user manual for more details.

Metering

Metering **(12)** comprises of two 6 LED bargraphs capable of measuring either audio output level or gain reduction.

The meter mode is set by the toggle switch **(13)** as follows;

Toggle up : sets the metering to measure audio output level.

Toggle down : sets the metering to measure gain reduction due to the action of the compressor.

Level metering

The level meter top red LED ‘0’ is calibrated to light at +18dBu. This is 10dB below the unit’s clipping level and is a common maximum level for many DAW audio interfaces. The level meter indicates level using the white scale ‘-21’ to ‘0’ in dBs.

It is important to note that the level meter follows the operational mode of the unit as follows;

When operating in either of the two L/R modes then the level meter will indicate left and right audio output levels.

When operating in Mid/Side then the level meter will read the internal audio levels of the MID (left hand bargraph) and SIDE (right hand bargraph) audio signals just before the final conversion into left/right output signals. When metering in this mode be aware that the meters will not be indicating the final left/right output levels.

Gain reduction metering

The gain reduction meter works as a top down meter to the orange scale marked from ‘1’ to ‘12’. Only one led lights at a time to show the amount of gain reduction taking place, in dBs. Gain reduction more than 12dBs will show as a solid on state of the green ‘12dB’ LED.

Internal module options

Calibration adjustment for the stereo width control

There is a small recessed screwdriver adjust potentiometer **(7)** which is used to calibrate the 12 o’clock position of the stereo width control **(1)**. This is factory preset and should not require user adjustment. Contact Safe Sound Audio should you think that the calibration needs to be reset and we will forward detailed instructions.

Calibration adjustment for L/M and R/S levels

There are two small recessed screwdriver adjust potentiometers **(3)** which can be used for fine level adjustment of the audio output levels from the unit. This is useful should you wish to correct a left/right level imbalance through your recording chain. These calibration controls are in circuit at all times and (unlike the main level controls) are never affected by the operation of the compressor in/out toggle switch.

The nominal calibration range for each control is ±3dB.

Setting transformers to low ‘clean’ or high ‘coloured’

It is possible to change the drive impedance to the transformers in order affect the coloration of the audio output signal. This can be very desirable to ‘warm up’ audio which has mostly be processed in the digital domain. It is necessary to remove the module from the rack frame in order to access the setting switch.

The location and setting of the miniature toggle switch is shown below.

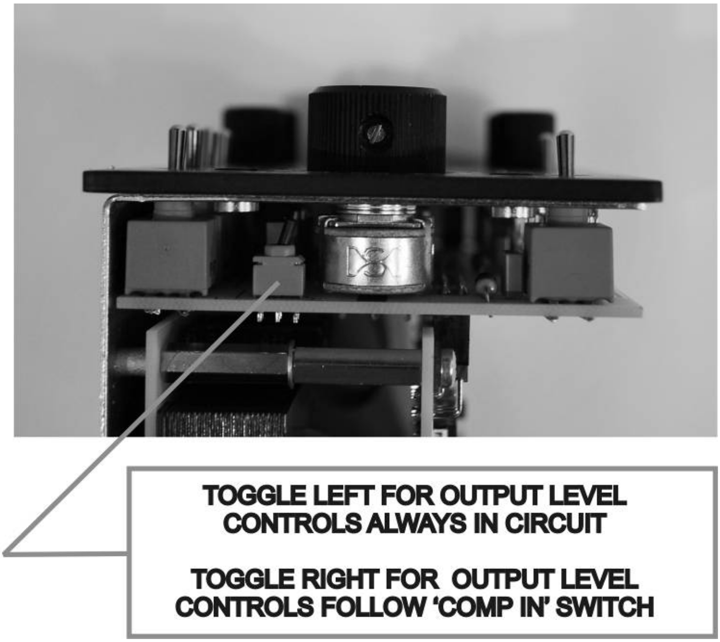
The factory setting is for high ‘coloured’

Switching the operation of the output level controls

It is possible to select whether the main output level controls **(10)** and **(14)** are in circuit at all times or are active only when the compressor is switched in. Linking their operation to the compressor allows easier comparison of compressed and non compressed material taking account of the gain make-up only being required when the compressor is active.

It is necessary to remove the module from the rack frame in order to access the setting switch.

The location and setting of the miniature toggle switch is shown below.



The factory setting is for the level controls to be switched in and out in tandem with the compressor

Understanding and using M/S (MID and SIDE) audio signals

Handling stereo audio as mid and side signals has been common for many years as a recording technique especially when doing live stereo recording using a pair of microphones. However the application of mid and side audio processing as a post production tool is now gaining popularity and offers a whole new palette of tonal shaping especially during the final mixdown stage of music production.

So what is MID and SIDE?

MID = LEFT + RIGHT
SIDE = LEFT - RIGHT

and conversely

LEFT = MID + SIDE
RIGHT = MID - SIDE

In normal stereo (think about two loudspeakers), the stereo audio signal is fed as left (to the LH speaker) and right (to the RH speaker). You sit listening in the central position and you hear a stereo representation of what is actually two discrete signals, one being sent to each speaker.

In M/S (which is still a representation of stereo audio), the two signals describe the centre (M or mid), and both sides together (S or sides).

The **mid** channel handles those elements which are mostly panned centre, whereas **side** handles these signals which are mostly panned away from centre (so side might be thought of as the stereo content of the audio).

Stereo width control is achieved by;

Less stereo width : boosting the level of the mid signal whilst simultaneously reducing the level of the side signal. At the extreme setting the side signal is reduced close to zero and you have a mono mix of the left and right audio appearing on the left and right outputs from the unit.

More width stereo width : boosting the level of the side signal whilst simultaneously reducing the level of the mid signal. At the extreme setting the mid signal is reduced close to zero and you have a very exaggerated wide stereo sound field appearing on the left and right outputs from the unit.

Compressing in mid/side allows you to compress central sounds (e.g. main vocals, kick drum and bass guitar) differently from side sounds (e.g. mic overheads, rhythm guitars, backing vocals).

So why compress and balance in MID/SIDE?
With the increasing track count available in today's DAWs, it's all too tempting to constantly tinker with individual track elements within the mix even during the final stages of mixdown, and before you know it, the coherence of the mix which you've been working so hard to achieve, has been lost.

Mid/Side processing allows you to work on the width, depth and dynamics of the stereo sound field during the latter stages of the mix and is especially useful when working with stereo stems (subgroups) which are then mixed to form the final stereo audio master.

<u>Let's take the example of a stereo drums and bass stem</u>
So you're pretty happy with the balance of the individual elements of the drums and bass but they sound too flat and dull within the overall mix. We can apply some gentle widening of the stereo image which will push the stereo overheads outwards. This will add a little depth and brightness without any overall level increase.

The stereo content (side) is better now but the drum/bass mix still sounds a bit cluttered. As the bass guitar and kick drum are usually panned to centre, we can use the on-board high pass filter to roll off some of the low frequency from the SIDE audio and that will create some more frequency space for the kick and bass.

Finally we can fine tune the levels of the drums/bass submix using the MID and SIDE output level controls.

And remember we haven't even touched the compressor controls yet!

Balancing lead and backing vocals

A typical vocal submix mix would place the lead vocal centre stage, maybe with double tracking processed with delay and reverb and panned right of centre to thicken up the lead vocal. Then a variety of backing vocals coming in on certain lines of the verses and in the choruses, panned left and right. You've got the idea.

So you get to the final mix and what you thought was thickening the vocals nicely has actually made them unintelligible and the backing singers sound like they're performing in another country - they're so far out of the mix!

Sometimes going back to the individual elements and changing levels, channel compression and reverb settings doesn't easily fix the problems within the overall mix.

Using M/S we can reduce the stereo width (of the vocals submix) a little to bring the backing singers back in from the cold.

Now let's compress the centre placed 'clean' lead vocal to bring it up in the mix a little without increasing the peak vocal level. That will gain us some intelligibility back without losing the thickening effect (from the doubled tracking in the SIDE signal). The side audio is left uncompressed as it's already heavily processed.

As before, the side audio, with its more heavily processed effects, is cluttering the mix; so we can apply some gentle side signal low frequency cut and create some more space for the centre vocal to sit within the mix.

There are many applications to compress and level balance within the MID/SIDE domain which allow fine tuning of the stereo width and depth of a mix during the mixdown stage of the production. This allows you to move forwards with the mix rather than a constant revisiting of the track elements and will help speed your workflow especially when dealing with very busy large track count mixes.

Technical Specification

STEREO TOOLBOX AUDIO PROCESSOR

Physical

Size :	API™ 500 Series dual module compatible (76mm wide by 133mm high by 150mm deep)
Weight :	1.14kg (2.5lbs)
Power requirements :	±16 volts dc at ±230mA to the standard API 500 Series pin-outs
Fusing :	On board ‘auto reset’ polyfuses protecting the ±16V rails

Main processor audio path

Frequency response :	-0.25dB at 10Hz, -0.5dB at 70kHz
Distortion :	on board selectable to low or high coloration
Low color :	0.06% up to 500Hz, < 0.02% at 1kHz and above
High color :	0.2% to 1% up to 500Hz, typically 0.04% at 1kHz and above
	Level and frequency dependent

Maximum input level

Line input :	+24dBu

Input impedance

Line input :	> 10k ohms

Output Noise

-90dBu RMS unweighted, measured 22Hz to 22kHz
-92dBu RMS A-weighted, measured 22Hz to 22kHz

Maximum Output Level (transformer balanced)

+28dBu at 1kHz and above, +24dBu at 50Hz (onset of transformer core saturation)

Output Gain (post compressor)	±15dB
Output impedance	50 ohms (low color) or 600ohms (high color)

Compressor

Threshold range :	+13dBu to -27dBu
Attack Time :	1ms or 10ms
Release Time :	programme related from 50ms to 1.5s
Ratio :	1:1 to 30:1
Gain reduction meter:	calibrated in dB

High Pass Filter

Sweepable :	17Hz to 2.1kz, 12dB per octave
Switchable :	to mono sidechain or SIDE audio path
Metering	Selectable to either level or gain reduction
Level metering :	LEFT/RIGHT or MID/SIDE depending on mode setting
Level metering :	0dB (red LED) calibrated at +18dBu
Gain Reduction metering :	measures from top of scale (red LED) downwards (1dB to 12dB+)

API is a registered trademark of Automated Processes Inc.

Warranty

Safe Sound Audio provides a 12 month warranty from the purchase date according to the following conditions;

This warranty is not transferable and applies only to the original purchaser of the unit.

If the unit should become faulty, then contact Safe Sound Audio or your local distributor for a Returns Authorisation Number. No items will be accepted for warranty repair without this authorisation number.

You must be able to produce proof of the purchase date.

If the returned unit should prove faulty then, at Safe Sound Audio's choice, we will either repair or replace the unit.

The customer is responsible for the cost of sending the unit back to Safe Sound Audio or our authorised distributor including insurance of the unit during shipping.

Safe Sound Audio or our authorised distributor will be responsible for the cost of shipping the repaired or replaced unit back to the customer including insurance of the unit during shipping.

The warranty will be void if the unit has;

- Suffered physical damage.
- Been repaired or modified by anyone other than Safe Sound Audio or its authorised representative.
- Has been connected to an incorrect source of power.
- Has been damaged due to liquid spillage.

Safe Sound Audio shall not be liable for any special or consequential damages resulting from the use of this product.

How to contact us

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Safe Sound Audio reserves the right to make changes and improvements to the design of this product without notice.