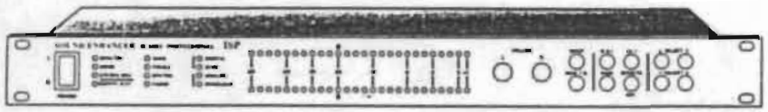
Professional Sound Enhancer 5022

User Manual



CONTENTS

1 General Description

1 Introduction

The Sound Enhancer is a unique combination of:

- Digital Sound Processor for de-clicking and performing a wide range of audio effects;
- A/D Converter (20-bit).
- D/A Converter (DAC-7 Bitstream).
- Sample Rate Converter (any valid digital signal between 15 and 50 kHz to either 44.1 kHz or 48 kHz).

The Sound Enhancer:

- is very easy to use and requires no special skills other than careful listening. It operates in real-time, so its influence can be immediately heard.
- Works in the digital domain, which means that the functions and features work precisely and do not introduce channel differences or audible noise.
- Is transparent for subcodes (see section 6.5).
- Can operate in a stand alone mode or in a studio environment. Slave mode operation is possible via either word clock or AES/EBU sync.

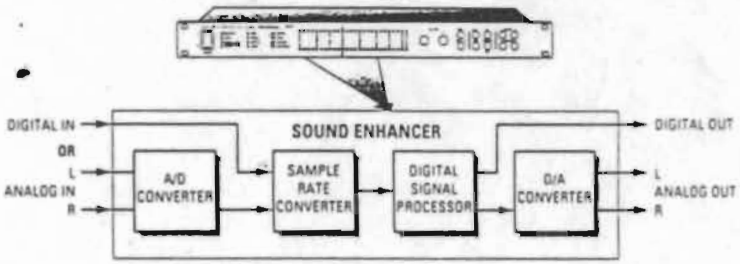


Figure 1.1: Shows an overall functional diagram of the Sound Enhancer.

2 Available functions and effects

The Sound Enhancer DSP provides the following audio functions:

- scratch suppression (de-clicking)
- noise reduction, offering a low-pass filter with an adjustable cut-off between 5.5 and 16 kHz.
- stereo enhancement, allowing a stereo effect to be created from a mono signal
- compression/expansion
- fader
- bass, treble, and volume control
- Quantization Noise Imaging shifts the quantization noise present in 16-bit digital signals outside the audible frequency range
- jitter removal is implicit in the handling of digital signals

- pitch variations (or varispeed) of up to +/- 12% can be handled without effecting the output sample rate
- slave mode using either word clock or AES/EBU sync signal
- digital level indication and control via dual LED bars.

For more details see section 6.4.

3 Two examples of applications

The following are just two of the many applications the Sound Enhancer can be used for:

- reducing the audibility of defects caused by scratches on gramophone records (de-clicking) or removal of other "sharp" defects (see figure 1.2);
- audio compression in order to change the dynamics of a recording to give a more convenient/suitable performance in noisy environments, e.g. classical music in a car.

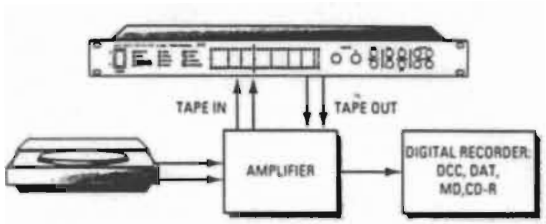


Figure 1.2: Typical application of Sound Enhancer.

4 About this manual

The Sound Enhancer is extremely versatile. The various configurations possible (input and output), combined with the available audio effects, are extensive. A complete description of all potential configurations and applications is well beyond the scope of this manual. Therefore some experimentation may be required for optimum results. However, the user is advised to follow this manual closely.

To get the best from the Sound Enhancer:

1. Read and follow the instructions in **Chapter 2** to unpack and turn the Sound Enhancer on.
2. Familiarise yourself with the controls, etc. in **Chapter 3**.
3. Install the Sound Enhancer [connect input(s) and output(s)] as described in **Chapter 4**.
4. For operation refer to **Chapter 5**.

Chapter 6 contains a technical description of the Sound Enhancer.

Reference is made to "Cinch connectors". In some countries this connector is known as an "RCA connector".

2 Unpacking

The Sound Enhancer IS5022 series is supplied in various different versions. Different mains adapters are supplied in order to accommodate the various mains requirements around the world. However, all models have the same operational characteristics.

1 Procedure

1. Remove the Sound Enhancer and other equipment from the box in which it is supplied. It is advisable to keep the box and packing for future use.
2. Check that the following equipment is included:
   - Sound Enhancer itself;
   - instruction manual (this document);
   - mains adapter;
3. Check that none of the equipment shows signs of being physically damaged.
4. Check that the mains voltage specified on the mains adapter is correct.

If any parts are damaged or missing, or the mains adapter voltage is wrong, report the details immediately to your supplier.

For ease of identification, report the 12 digit code number on the under side of the Sound Enhancer and on the shipping carton. Code number = 9022 67502 2/XX (XX = coding for configuration, voltage and mains frequency). 01 = Europe; 02 = United Kingdom; 03 = UL (USA). For further information refer to Chapter 6.

2 Quick functional check

Connect power to the Sound Enhancer as follows. (Refer to figures 3.1 and 3.2 for locations):

1. Ensure that the POWER switch (F1) on the front panel is switched off.
2. Connect the power adapter cable with the DIN connector to the POWER SUPPLY DC IN (B9) socket on the back panel of the Sound Enhancer.

Figure 2.1: Connecting the power adapter.

Note: The arrow on the connector should be uppermost (see figure 2.1). Indentations must match with the insert of the receptacle.

3. Connect the mains cable of the power adapter to a mains socket of the correct voltage (see sticker on mains adapter).
4. Set the POWER switch (F1) on the front panel of the Sound Enhancer to on.

At first a number of LEDs may come on and then off. After a second or so, only the left-most LEDs in the LED bars and one of the input LEDs should be on. If, after making the power connections in the proper manner, no LEDs come on then it is likely that the Sound Enhancer has been damaged during shipping.

3 Controls, Indicators & Connectors

This chapter describes the controls, indicators, and connectors located on the front and back panels of the Sound Enhancer.

1 Front panel

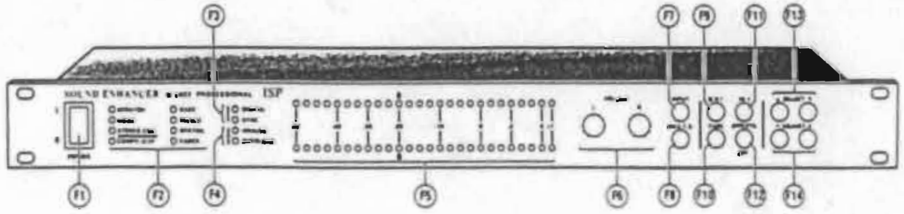


Figure 3.1: Sound Enhancer front panel.

The front panel is shown in figure 3.1. It is divided into four main functional blocks. They are, from the left:

- the power switch
- the effect and input LEDs;
- two LED bars;
- the controls.

These are described in the following sections. For references in brackets, e.g. (F1), see figure 3.1.

Power switch
The power switch (F1) is a simple rocker switch which connects the 9 V d.c. from the mains adapter to the Sound Enhancer electronics.

Effect and input LEDs
The effect LEDs (F2) are: SCRATCH, NOISE, STEREO ENHANCEMENT, COMPRESS/EXPAND, BASS, TREBLE, SPATIAL, and FADER:

- When an effect is not active, the LED is off.
- When the LED flashes the setting can be adjusted.
- When the effect is active the LED is on.

The input LEDs, DIGITAL (F3) and ANALOG (F4), indicate which input is selected.

The SYNC LED (F3) comes on when the digital input signal is synchronised with the signal at the word clock connector on the back panel. If there is a valid digital signal at the selected digital input (XLR or Cinch connector), and its sample rate is the same as the selected output sample rate synchronisation will.occur. This is indicated by the SYNC LED flashing in an isophase mode.

The OVERLOAD LED (F4) comes on when the analog input signal level is too high. This can be corrected by adjusting the VOLUME controls (F6) (see below), or the output level of the source device.

The LED bars
The central part of the front panel contains two horizontal LED bars (F5) with a range from -60dB to +1dB with green/orange/red graduations. During normal operation, the LED bars display the audio input signal level. The upper row is the left channel and the lower row is the right channel.

While adjustments are being made to the effects, the LED bars indicate the amount (or level) of effect being applied, as set by the ADJUST buttons. In this role the dB scale is redundant.

Operator controls
The right hand part of the front panel contains the operator controls:

- VOLUME (F6) comprises two (left and right) potmeters that adjust the level of the audio inputs, and thus the balance. The potmeters can be disabled (see section 4.10).
- INPUT (F7) is a toggle action push button switch that alternately selects the digital and analog input. The selected input is indicated by either the DIGITAL (F3) or ANALOG (F4) LED being on. In order to avoid accidental switching to another input, the button must be pressed for about a second before the switch is made.
- MAX T.S. (F8) is a two position toggle action push button switch that selects the maximum level of scratch suppression (TS = transient suppression). The scratch effect must be activated. When TS is on, it is indicated by an LED in the switch coming on. MAX T.S. should only be used in extreme situations, or with mono signals, because its effect is to virtually reduce stereo to mono. To compensate for this, MAX TS can be used in combination with the stereo enhancement effect.

The next group comprises four yellow toggle action push button switches. Each of these buttons contains an LED which comes on when the switch is active.

- Q.N.I. (F9) activates/de-activates the Quantization Noise imaging effect.
- FADE (F10) activates the fade effect (when it is enabled, indicated by the FADER LED being on). The button sequentially activates fade in and fade out, so care should be taken when using it.
- 44.1 (F11) sets the output sample rate to 44.1 kHz instead of the default 48 kHz. Pressing the switch a second time reverts the output sample rate back to 48 kHz. In order to avoid noise during switching, the change over is implemented via an automatic "fade out", change of sample rate, "fade in" sequence. Consequently switching between sampe rates is not instantaneous. The user is advised to select the sample rate prior to applying the other effects and functions. When the LED in the push button is off, the sample rate is48 kHz.

- EFFECTS OFF (F12) switches out the active effects. It can be used to make a comparison with the raw audio, and thus hear the influence of the active effects and settings. When the button is pressed a second time the effects become active again. If used while an effect LED is flashing, only that effect will be toggled off.

  Note: *If the Sound Enhancer is switched off while the effects are switched off, the active effects and settings are lost when it is next switched on.*

The next group of switches is used to select, adjust, and activate the effects.

- SELECT (F13) comprises two white push button switches that are used to sequentially step through the effects:

  SCRATCH
  NOISE
  STEREO ENHANCEMENT
  COMPRESS/EXPAND
  BASS
  TREBLE
  SPATIAL
  FADER

  The '+' switch steps forward through the effects and the '-' switch steps backwards. The selected effect is indicated by its LED flashing and the LED bar will indicate the level of adjustment.
  Once the SELECT button is touched, there is a six second period to start adjusting the setting (see below). If no adjustment is made the LED bar will revert to indicating the audio level.

- ADJUST (F14) comprises two red push button switches that are used to set the level for the selected effect (indicated by the LED flashing). When either of these switches is pressed the selected effect will be activated with the set adjustment.

  The '+' switch increases the effect level and the '-' switch decreases it. The effect level is indicated on the LED bars.

  When no further adjustment is made for six seconds, the effect LED stops flashing and stays on.

2 Rear panel

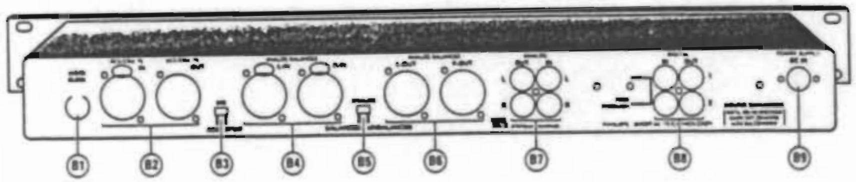


Figure 3.2: Sound Enhancer rear panel.

The rear panel contains the input and output connectors. For references in brackets, e.g. (B1), see figure 3.2.

- WORD CLOCK (B1) is a BNC connector to which a word clock signal can be connected for use instead of the Sound Enhancer's internal clock.
- AES/EBU IN and OUT (B2) are two XLR connectors. The digital input signal must conform with IEC 958. Any sample rate frequency between 15 and 50 kHz can be used. No adjustment of the Sound Enhancer is necessary for different input sample rates. The digital output signal conforms with IEC 958. The output sample rate frequency is selected by a switch on the front panel (F11) between 44.1 and and 48 kHz.
- DIG AES/SPDIF (B3) is a two position manual slider switch used to select either the AES/EBU (B2) input, or the SPDIF input (B8), as the digital input selectable by the INPUT (F7) switch on the front panel.
- ANALOG BALANCED L-IN & R-IN (B4) are two XLR connectors for the input of balanced analog line-in stereo signals.
- ANALOG BALANCED/UNBALANCED (B5) is a two position manual slider switch used to select either the analog balanced (B4) input, or the unbalanced input (B7), and the analog input is selectable by the INPUT (F7) switch on the front panel.
- ANALOG BALANCED L-OUT & R-OUT (B6) are two XLR connectors for the output of balanced analog line-out stereo signals.
- ANALOG OUT IN (B7) are four Cinch connectors for the input and output of unbalanced analog line-in and line-out stereo signals.
- DIGITAL IN OUT (B8) are four Cinch connectors for the input and output of SPDIF digital signals. Two are used as output; the other two are tied together. This means that one pair can be used as input while the second pair can be used as a feed-through where two Sound Enhancers are being used together.
- POWER SUPPLY DC IN (B9) is for the power input from the mains adapter.

4 Installation

1 Introduction

The Sound Enhancer can be used in many different roles and there are many ways to connect the unit. This chapter lists some, (NOT ALL) of the most frequently used applications and related installations. Some experimentation may be necessary.

The Digital Signal Processor in the Sound Enhancer is controlled via the SELECT switches. However, three principal functions can be performed without using the SELECT switches. These are:

- A/D conversion;
- D/A conversion;
- Sample rate conversion.

The function is defined by the connected input(s) and output(s). The possible input/output combinations are:

- **analog/analog** the Sound Enhancer acts as an A/D and D/A converter.
- **analog/digital** the Sound Enhancer acts as an A/D converter;
- **digital/analog** the Sound Enhancer acts as a D/A converter;
- **digital/digital** the Sound Enhancer can act as a Sample Rate Converter, depending on the sample rate of the input signal and the selected output sample rate.

These functions are implicit and cannot be disabled by the user.

Notes:

1. *The mains adapter contains a transformer. In the event that unwanted influence on the audio performance of other equipment is observed it may be necessary to relocate the mains adapter.*
2. *An input signal can be sourced from a wide range of analog and digital equipment. Refer to the manual of the equipment concerned to see if the signal at a particular connector is suitable (see section 6.2).*
3. *The Sound Enhancer has no phono amplifier on board, so the phono output of a record player should not be connected directly to the ANALOG IN socket. Instead it must be connected to the PHONO IN connector of an amplifier for RIAA correction.*

CAUTION!!	*Cables should only be connected to, or disconnected from, the Sound Enhancer when it is switched OFF.*

2 Sound Enhancer as A/D Converter

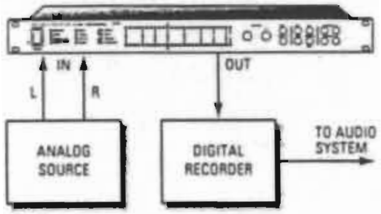


Figure 4.1: Sound Enhancer as A/D converter.

1. If necessary, switch the digital output signal to the required sample rate (44.1 switch).
2. Connect the output signals from the analog source to the required analog input connectors (balanced or unbalanced) of the Sound Enhancer.
3. Select balanced or unbalanced, as required, using the back panel switch on the Sound Enhancer.
4. Connect the required digital output (AES/EBU or SPDIF) of the Sound Enhancer to an appropriate digital input socket on the receiving device.
5. Switch the Sound Enhancer on.
6. Press the INPUT button to select the analog input, if necessary.
7. To adjust the input level turn the VOLUME rotary controls until the required level is obtained.

The Sound Enhancer is now configured as an A to D converter.

Refer to Chapter 5 for details about how to apply the audio effects, if required.

3 Sound Enhancer as D/A Converter

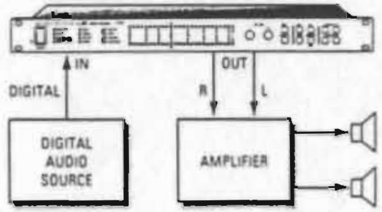


Figure 4.2: Sound Enhancer as D/A converter.

1. Connect the digital output signal from the audio source to either the AES/EBU IN or DIGITAL IN connector of the Sound Enhancer.
2. Select AES/SPDIF, as required, using the back panel switch on the Sound Enhancer.
3. Connect the required analog output of the Sound Enhancer (balanced or unbalanced) to the analog input connector of the receiving device.
4. Switch the Sound Enhancer on, if necessary.
5. Press the INPUT button to select the digital input.
6. To adjust the input level, turn the VOLUME rotary controls until the required level is obtained.

The Sound Enhancer is now configured as a D to A converter.

Refer to Chapter 5 for details about how to apply the audio effects, if required.

4 Sound Enhancer as Sample Rate Converter

Different digital devices use different sample rates: CD and CD-R use 44.1 kHz while R-DAT uses 48 kHz. Generally speaking, 44.1 kHz is used in most domestic digital recording applications, and 48 kHz in most (broadcast) studio applications.

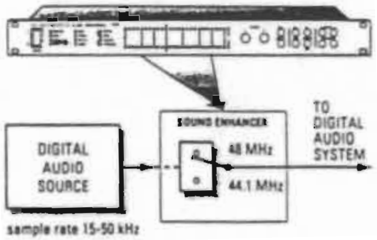


Figure 4.3: Sound Enhancer as sample rate converter.

To convert the frequency of any digital signal with sample rate between 15 and 50 kHz to 44.1 kHz or 48 kHz, proceed as follows.

1. If necessary, switch the digital output signal to the required sample rate (44.1 switch).
2. Connect the digital output signal from the audio source to either the AES/EBU IN or DIGITAL IN connector of the Sound Enhancer.
3. Select AES/SPDIF, as required, using the back panel switch on the Sound Enhancer.
4. Switch the Sound Enhancer on.
5. Press the INPUT button to select the digital input, if necessary.
6. To adjust the input level turn the VOLUME rotary controls until the required level is obtained.

The Sound Enhancer is now configured as a sample rate converter.

Refer to Chapter 5 for details about how to apply the audio effects, if required.

5 Ampilifier based installations

For many applications it is practical to use an audio amplifier as a control centre or switch box (see figure 4.4).

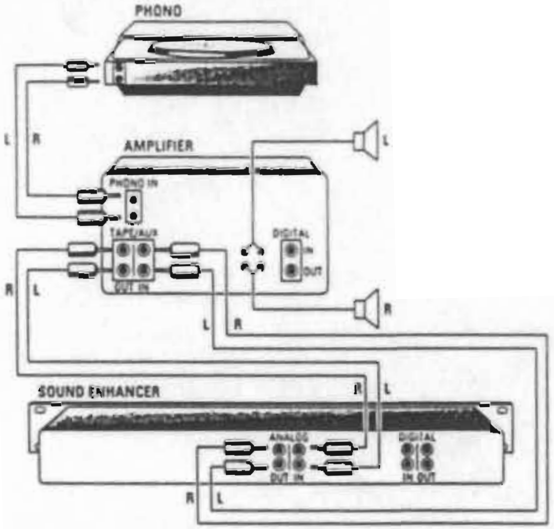


Figure 4.4: Audio ampilifier as a control centre or switch box.

Some amplifiers do not provide sufficient switching options. In such cases awkward rewiring can be avoided by including a cassette deck loaded with a blank cassette and switched to RECORD/PAUSE mode. Figures 4.5, 4.6, and 4.7 show various configurations how this can be achieved.

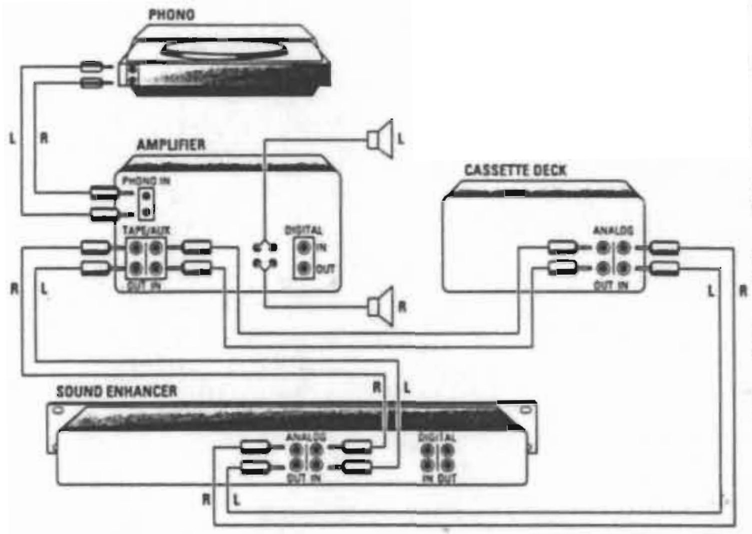


Figure 4.5: How to overcome insufficient switching options.

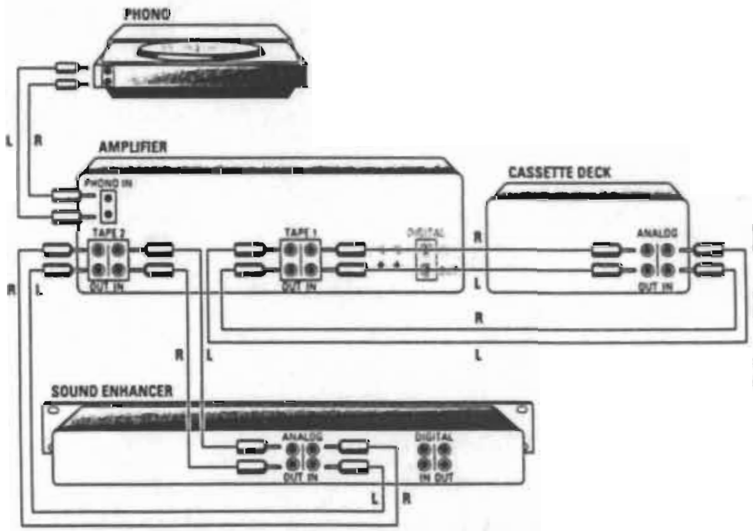


Figure 4.6: Installation for amplifiers with dual tape inputs/outputs.

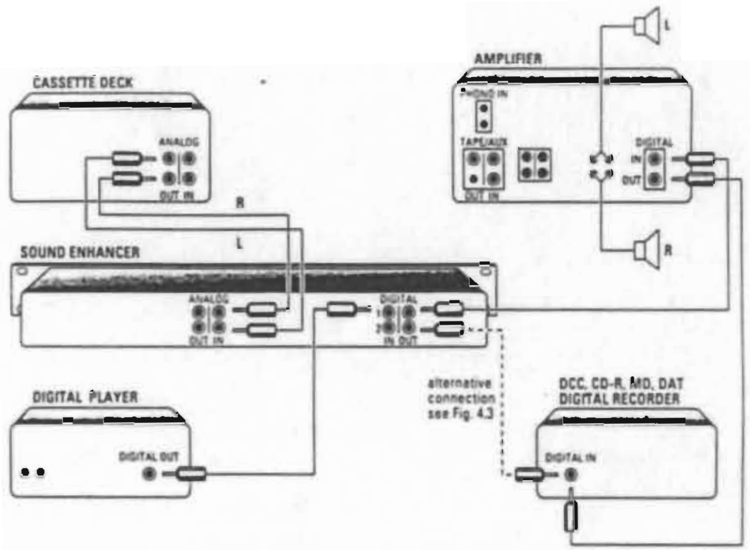


Figure 4.7: Direct connections and parallel inputs.

6 RECORDING AND PLAYBACK WITHOUT RECABLING

The Sound Enhancer's two digital inputs and outputs allow it to be used for both recording and playback without reconfiguring the audio cables. The signal paths are shown in figure 4.8.

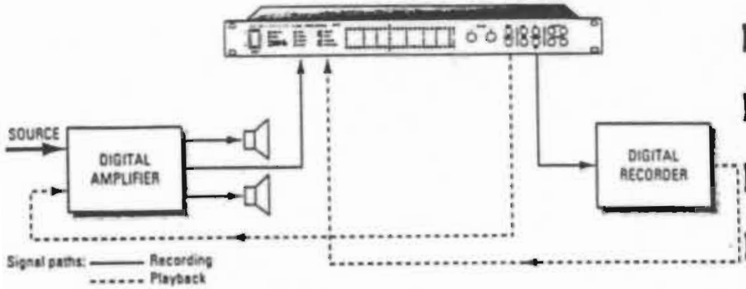


Figure 4.8: Configuration for recording and playback.

7 POST PROCESSING

Not all effects can be performed simultaneously, for example scratch suppression cannot be used simultaneously with compression/expansion (see section 5.2.1). This can be overcome by post-processing (which requires an interim recording). For this two digital recorders are necessary. The signal paths are shown in figure 4.9.

Note: The amplifier is only used for signal routing.

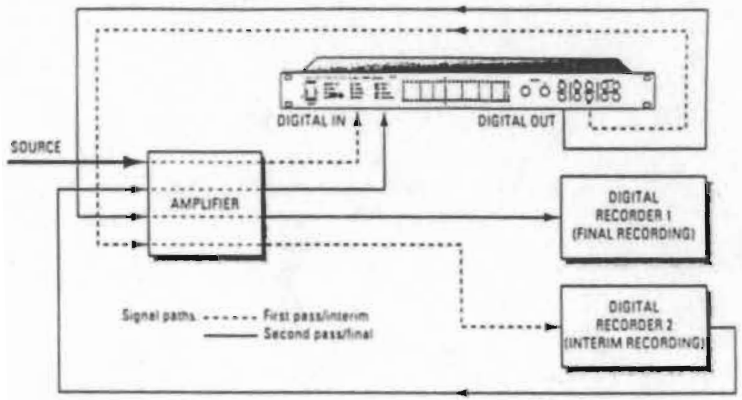


Figure 4.9: Sound Enhancer in post-processing configuration.

4 INSTALLATION

8 Slave mode

The Sound Enhancer can act as a slave. In this mode an external clock signal is used instead of the internal clock. There are two sources of external signal:

- a signal at the word clock connector
- a valid AES/EBU digital signal at either the XLR or SPDIF connector.

While the Sound Enhancer is slaved to the word clock the SYNC LED is permanently on.

While the Sound Enhancer is slaved to a valid AES/EBU input signal the SYNC LED flashes.

When no word clock is present and also no digital input signal, the Sound Enhancer is not in slave mode and uses its internal clock. The SYNC LED is then off.

The selected output sample rate (44.1 switch) can effect slave mode. The tables below show what the Sound Enhancer will slave to under different conditions (FS = sample frequency; External FS = word clock; source FS = AES/EBU input signal).

Mode of operation	44.1 switch	Source NONE	Source CD 44.1	Source DAT 48
No external FS connected	44.1	No slaving	Slaving on source FS	No slaving
	48	No slaving	No slaving	Slaving on source FS
External FS connected (FS = 44.1)	44.1	Slaving on external FS	Slaving on external FS	Slaving on external FS
	48	No slaving	No slaving	No slaving
External FS connected (FS = 48)	44.1	No slaving	No slaving	No slaving
	48	Slaving on external FS	Slaving on external FS	Slaving on external FS

9 Calibration

The Sound Enhancer is factory calibrated for an analog input signal of 15 dBu. This means that when such a signal is applied, and the potmeters turned up to maximum, a 0 dB Fs (full scale) signal is present at the digital output. This ensures that the full 20-bit digital range of the sample rate converter is used.

If an analog signal greater, or less, than 15 dBu is to be applied as the reference then the Sound Enhancer must be recalibrated for that signal. To do this proceed as follows:

1. If it is on, switch the Sound Enhancer off.
2. Turn both the potmeters to the maximum position (fully clockwise).
3. Remove the top panel of the Sound Enhancer.
4. Locate the XLR PCB (see figure 4.10)
5. Locate the two trim pots 3247 and 3248 (see figure 4.10).
6. Apply an analog signal of the required level to the analog input.
7. Connect a suitable digital signal measuring device (such as Audio Precision System One) to either digital output connector.
8. Switch the Sound Enhancer on.
9. Adjust the trimpots so that both output channels (L and R) give a reading of 0 dB. (Alternatively obtain the same dBu level at the analog output as the applied input signal.)
10. Switch the Sound Enhancer off; disconnect the measuring device, replace the top panel.

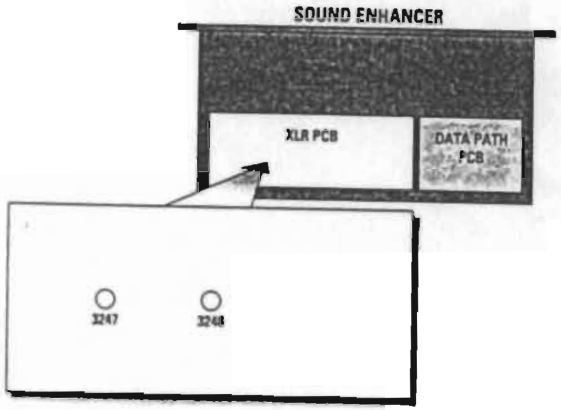


Figure 4.10: Location of trimpots.

10 Disabling the potmeters

To avoid the possibilty of the potmeters being inadvertently adjusted, they can be disabled.

Note: *When the potmeters are disabled, there is no control over the level of the analog input signal. For this reason the Sound Enhancer should be calibrated for the applied analog input signal. See the previous section.*

To disable the potmeters proceed as follows:
1. If it is on, switch the Sound Enhancer off.
2. Remove the top panel of the Sound Enhancer.
3. Locate the XLR PCB (see figure 4.11)
5. Locate the two jumpers 1206 and 1207.
6. Place both jumpers in the alternative position.
7. Replace the top panel.

The potmeters are now disabled. To enable them again simply replace the jumpers to the original positions.

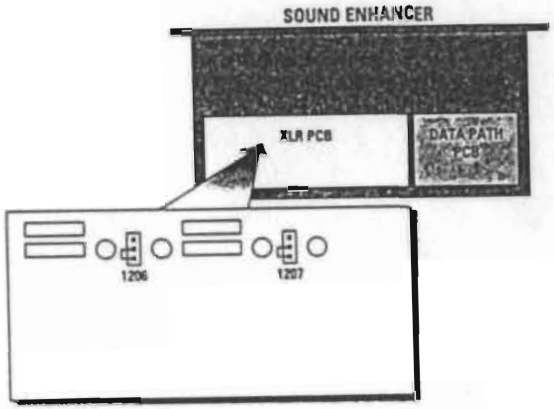


Figure 4.11 Location of jumpers (factory settings shown).

5 Operation

1 Setting to work

It is important to adhere to the following sequence:
1. Connect the input(s) and output(s) . See Chapter 4.
2. Set the volume control (potmeters) to the mid position to avoid excessive volume when the analog input signal is applied.
3. Switch the unit on (POWER switch).
4. Select input (INPUT switch)
5. Select the required output sample rate. Default is 48 kHz, if 44.1 kHz is required press the 44.1 button on the front panel.

Any required audio effects can now be applied. See the following section.

Note: *Effects do not have to be used. The Sound Enhancer is able to perform as an A/D or D/A converter or as a sample rate converter without any audio effects.*

2 Audio effects

Once the input(s) and output(s) have been connected for the required function (see Chapter 4), any required effects can be activated.

Note: *In order to obtain the optimal setting, the Sound Enhancer should be configured so that the various effects can be heard by means of headphones or speakers .*

The SELECT buttons (+ and -) sequentially select the following audio effects:
- scratch (suppression)
- noise filter
- stereo enhancement
- compression/expansion
- bass
- treble
- spatial
- fader
- volume setting

(Quantization Noise imaging is handled differently and is described separately in section 5.3.)

The following is a general description of how an effect is activated and adjusted. (Specific descriptions for each effect are given in the following sections of this chapter.)

1. Press the SELECT button until the LED of the required effect flashes. The current setting for the effect is indicated on the LED bar.

2. Obtain the required setting by listening carefully and pressing the + and - ADJUST buttons. The LED bar will show the setting. The act of making an adjustment means the effect **will** be activated. If the ADJUST buttons are not touched, the LED stops flashing after six seconds and the effect is not activated.
3. To hear the influence of the effect, use the EFFECT ON/OFF button to toggle it on/off while the LED is flashing. (If the EFFECT ON/OFF button is toggled while no LED is flashing, all the activated effects will be toggled on/off.)
4. When the effect is satisfactory, do not touch the ADJUST buttons for six seconds and the effect is activated. This is indicated by the LED stopping flashing and remaining on. The LED bar reverts to displaying the audio level.

2.1 Combinations of effects

The Sound Enhancer is able to perform either:
- scratch and/or noise filter and/or stereo enhancement; **or**
- compress/expand.

Bass, treble, spatial and fader can be used with any of these effects (refer to figure 5.1).

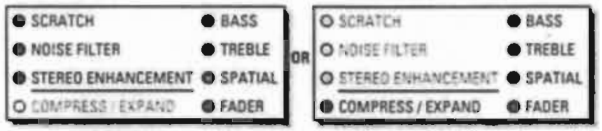


Figure 5.1 Valid combinations of effects

2.2 Activating the effects

This section describes how to activate the different effects.

2.2.1 Scratch, noise filter, stereo enhancement

Note: *These functions cannot be used while compress/expand is active.*

To activate either scratch, noise filter or stereo enhancement.
1. Repeatedly press either of the SELECT buttons until the LED of the required effect (scratch/noise filter/stereo enhancement) flashes. (If the compress/expand effect was active, it will be automatically de-activated.)
2. Press the ADJUST buttons to obtain the required setting. This is best determined by experiment, i.e. adjust the setting, listen, re-adjusting the setting, and so on. When the ADJUST buttons are not touched for six seconds the LED will stop flashing and remain on.

The scratch suppression level should be set as low as possible for an optimum result. This should be with only green LEDs on. If the level is set so that the yellow LEDs are on, then caution should be exercised. If the level is set so that either or both of the red LEDs are on, the music will be audibly impaired. The two marks above and below the LED bar are the suggested default value. To temporarily activate maximum scratch suppression, press the MAX T.S. switch.

The record player should be fitted with a good quality stylus and cartridge. The performance of the Sound Enhancer as a scratch suppressor is dependent on the quality of the cartridge. The scratch suppression is also dependent on the stylus force, too low or high will give less than optimum results. Similarly, lateral pressure should also be equal.

Sometimes a recording has only one or two bad clicks. In this case, set the lowest optimum level but increase it during the period that contains the bad click(s).

For noise filter, the LED bar range is from about 6 kHz (all LEDs on) to about 16 kHz (two LEDs on). Refer to chapter 6.

2.2.2 Compression/expansion

Note: *This function cannot be used while any of scratch/noise filter/stereo enhancement effects is active.*

To activate compression/expansion:
1. Repeatedly press either of the SELECT buttons until the compress/expand LED flashes. (If any of the scratch/noise filter/stereo enhancement effects were active, they will be automatically de-activated).
2. Press the ADJUST buttons to obtain the required setting.
3. When the ADJUST buttons are not touched for six seconds the LED will stop flashing and remain on.

Note: *Using compression/expansion has a distinctly audible influence on the volume. If expansion is applied, the volume will decrease; if compression is applied, the volume will increase. Adjust the volume level, as required, at the device connected to the output of the Sound Enhancer.*

2.2.3 Bass and treble

The bass and treble effects can be used individually or in any combination. They are activated as follows:
1. Repeatedly press either of the SELECT buttons until the LED of the required effect (bass or treble) flashes.
2. Press the ADJUST buttons to obtain the required setting.
3. Do not touch the ADJUST buttons for six seconds and the LED will stop flashing and remain on.

2.2.4 Spatial

Spatial can widen or narrow the sound stage in three steps between "Stereo" and "Mono". The default setting is "normal" stereo. The spatial effect can be used alone or in any combination with other effects. It is activated as follows:

1. Repeatedly press either of the SELECT buttons until the SPATIAL LED flashes.
2. Press the ADJUST buttons to obtain the required setting.
3. When the ADJUST buttons are not touched for six seconds the LED will stop flashing and remain on.

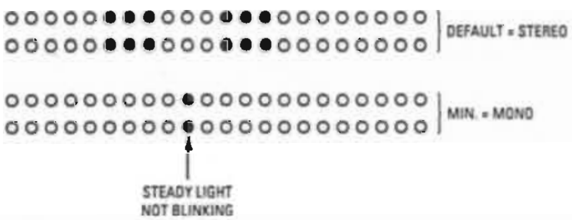


Figure 5.2 LED bar for spatial effect

2.2.5 Fader

The fade effect works in a slightly different way from the other effects. The fader is enabled, and the fade time set, as follows:

1. Repeatedly press either of the SELECT buttons until the Fader LED flashes.
2. Press the ADJUST buttons to obtain the required fade time. The LED bar range is from 0.5 seconds (minimum) to 11 seconds. The default setting is zero seconds (zero fade time gives immediate muting.
3. Do not touch the ADJUST buttons for six seconds and the LED will stop flashing and remain on.

The fade effect can now be activated by pressing the FADE button.

Note: The FADE button has a toggle action for fade in/fade out, so care should be taken when using it so that the required action is obtained.

2.2.6 Volume setting

During normal operation the LED bar shows the audio levels (L and R). The input levels can be individually adjusted via the two VOLUME pot. meters. The moment either of these pot meters is adjusted, the LED bar will indicate the position of the potmeters. After two seconds without adjustment, the display will revert to showing the actual input signal.

2.3 De-activating effects

This section applies to all effects except the fader (see previous section) and Q.N.I. (see next section).

If none of the effects are active, it is known as defeat mode. Effects can be de-activated as described below.

To de-activate all the effects (except Q.N.I.):

1. Ensure no effect LEDs are flashing.
2. Press the EFFECTS OFF button.

All the active effects will be de-activated (LED(s) off).

Where a number of effects are active and you want to de-activate only one of them (not Q.N.I.):

1. Use the SELECT buttons to select the effect you wish to de-activate.
2. While the LED is flashing, press the EFFECTS OFF button.

The effect will be de-activated (LED off).

De-activated effects can be re-activated by pressing the EFFECTS OFF button again as long as the selected effect is flashing. The button has a toggle action.

3 Quantization noise imaging

Quantization Noise Imaging (Q.N.I.) can be used irrespective of the other effects.

Q.N.I. improves the audio quality of digital audio source material by moving the quantization noise present in the digital signal to outside the audible frequency range. The main advantage is that the audio performance, especially of low level signals/recordings, can be improved. Q.N.I. works for 20-bit and 18-bit recordings, but the benefit will be more noticeable for 16-bit recordings, e.g. CD. For analog recordings, or 20-bit digital recordings, the Q.N.I. effect will give no benefit.

To activate Quantization Noise Imaging:

1. Press the Q.N.I. button.

The Quantization Noise Imaging LED in the push button comes on. Q.N.I. has no adjustment, it is either on or off. The Q.N.I. button has a toggle action, so pressing it a second time cancels Q.N.I. If Q.N.I. is inadvertently left on, it will not have any negative influence on the audio output.

6 TECHNICAL DATA

1 GENERAL

Weight:

Sound Enhancer:	4.1 kg
Mains Adapter:	1.3 kg

Dimensions: mm	w	x	d	x h
Sound Enhancer:	(19")	x	290	x (1E)
Mains Adapter:	60	x	95	x 70

Power:

The Sound Enhancer is powered by d.c. It is supplied with a specific adapter for the local mains.

Type:	/01 Europe, /02 UK, /03 USA (UL)
Mains voltage:	230 V ±15%, 120 V ±15%
Mains frequency:	50 Hz, 60 Hz
Consumption:	approx. 25 W

Inputs:

Digital :	AES/EBU & SPDIF (complying with IEC 958) 15-50 kHz sample rate. Connector: XLR/Cinch
Analog 1:	Line-in, Connector: Cinch L/R (RCA)
Analog 2:	Balanced in, Audio Connector: XLR

Outputs:

Digital:	AES/EBU, Connector: XLR SPDIF (complying with IEC 958) Connector: Cinch/RCA 48/44.1 kHz sample rate switchable on front panel
Analog:	Line-out stereo (L/R) Connector: Cinch/RCA Balanced out (L/R), Connector: XLR

2 PERFORMANCE

Except where indicated as otherwise, all values are nominal or typical.

DIGITAL

20 bits audio-performance

Balanced:	XLR connectors
Unbalanced:	Cinch connectors

Input frequency:	15-50 kHz
Output frequency:	44.1 kHz or 48 kHz

Word clock

Input impedance:	45 kΩ
Input level:	min. 2 Vtt, max 6 Vtt
Frequency:	44.1 kHz or 48 kHz (tolerance: ±100 ppm)

ANALOG

Balanced (XLR)

Input sensitivity:	Adjustable: +14 dBm - 23.7 dBm (typ.+15dBm ±0.5 dBm) for maximum level (0 dB digital) Definition: 0 dBm = 0.775 V rms (1mW in 600 E)
Input Impedance:	46 kΩ
Output level:	15 dBm ±0.5 dBm for maximum level (0 dB digital)
Output Impedance:	48 Ω

***Note:** The Sound Enhancer is delivered calibrated for an input sensitivity and output level of 15 dBm.*

Unbalanced (Cinch)

Input sensitivity:	0.35 V rms for maximum level (0 dB dig) 55.5 mV rms for reference level (-16 dB digital) <u>1.2 Vp maximum input voltage!</u>
Input Impedance:	50 kΩ
Output level:	2 V rms for maximum level (0 dB digital) 0.32 V rms for reference level (-16 dB digital)
Output Impedance:	200 Ω

Balanced (XLR)

Analog in to analog out (complete analog path)

Unbalance:	max. 2 dB
Amplitude linearity:	max. ±0.15 dB (20 Hz - 20 kHz)
Phase linearity:	typ. ±5.8 ° (20 Hz - 20 kHz)
S/N ratio:	typ. 90 dB (Bandwidth 20 kHz) typ. 94 dB (A weighted)
Dynamic Range:	typ. 90 dB (1 kHz)
THD + Noise:	typ. 86 dB (1 kHz)
Outband Attenuation	60 dB above 25 kHz
Channel separation:	typ. 98 dB (1 kHz)
Low level linearity:	within 3 dB (at -90 dB)

Analog In

Unbalance:	max. 1 dB (adjusted)
Amplitude linearity:	max. ±0.05 dB (20 kHz - 20 kHz)
Phase linearity:	typ. ±4.5° (20 Hz - 20 kHz)
S/N ratio:	typ. 94 dB (Bandwidth 20 kHz) typ. 98 dB (A weighted)
Dynamic Range:	93 dB (1 kHz)
THD + Noise:	90 dB (1 kHz)
Outband attenuation:	60 dB above 25 kHz
Channel separation:	typ. 100 dB (1kHz)
Low level linearity:	within 1.5 dB (at -90 dB)

Analog Out

Output Voltage:	+15 dBm ±0.5 dB
Unbalance:	max. 1 dB
Amplitude linearity:	max ±1 dB (20 Hz - 20 kHz)
Phase linearity:	typ. ±1.3 ° (20 Hz - 20 kHz)
S/N ratio:	typ. 95 dB (Bandwidth 20 kHz) typ. 100 dB (A-weighted)
Dynamic range:	typ. 95 dB (1 kHz)
THD + Noise:	typ. 95 dB (1 kHz)
Outband attenuation:	60 dB above 25 kHz
Channel separation:	typ. 105 dB (1 kHz)
Low level linearity:	within 1.5 dB (at -90 dB)

Unbalanced (Cinch/RCA)

Analog in to analog out (complete analog path)

Unbalance:	max. 1.5 dB
Amplitude linearity:	max ±0.15 dB (20 Hz - 20 kHz)
Phase linearity:	typ. ±3.25° (20 Hz - 20 kHz)
S/N ratio:	typ. 93 dB (Bandwidth 20 kHz) typ. 96 dB (A-weighted)
Dynamic range:	typ. 92 dB (1 kHz)
THD + Noise:	typ. 88 dB (1 kHz)
Outband attenuation:	60 dB above 25 kHz
Channel separation:	typ. 100 dB (1 kHz)
Low level linearity:	within 3 dB (at -90 dB)

Analog In

Unbalance:	max. 0.8 dB
Amplitude linearity:	max ±0.03 dB (20 Hz - 20 kHz)
Phase linearity:	typ. ±2.75 ° (20 Hz - 20 kHz)
S/N ratio:	typ. 95 dB (Bandwidth 20 kHz) typ. 100 dB (A-weighted)
Dynamic range:	typ. 93 dB (1 kHz)
THD + Noise:	typ. 90 dB (1 kHz)
Outband attenuation:	60 dB above 25 kHz
Channel separation:	typ. 105 dB (1 kHz)
Low level linearity:	within 1.5 dB (at -90 dB)

Analog Out

Output voltage	2 V rms ±2.5 dB
Unbalance:	max. 0.5 dB
Amplitude linearity:	max ±0.1 dB (20 Hz - 20 kHz)
Phase linearity:	typ. ±0.5 ° (20 Hz - 20 kHz)
S/N ratio:	typ. 98 dB (Bandwidth 20 kHz) typ. 103 dB (A-weighted)
Dynamic range:	typ. 95 dB (1 kHz)
THD + Noise:	typ. 95 dB (1 kHz)
Outband attenuation:	60 dB above 25 kHz
Channel separation:	typ. 110 dB (1 kHz)
Low level linearity:	within 1.5 dB (at -90 dB)

3 Key components

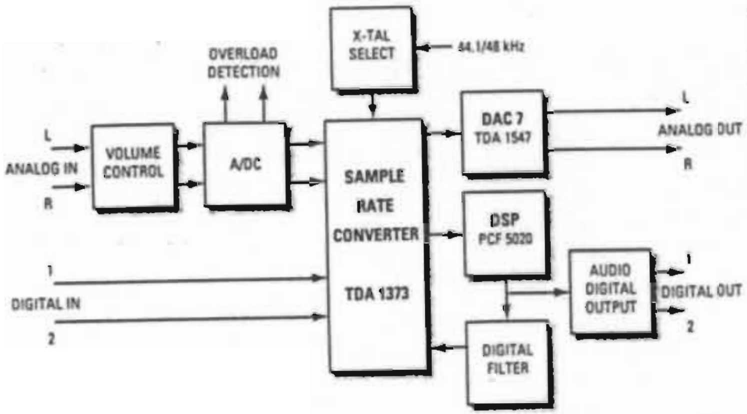


Figure 6.1: Sound Enhancer block diagram.

A/D Converter	AD MOD 79
D/A Converter	TDA 1547
DSP function	PCI 5020D
ADOC function	CS 8402
ADIC function	TDA 1373

4 Functional data

- **Sample rate conversion**
  The Sound Enhancer is equipped with a TDA 1373 Sample Rate Converter, which supports 20-bits. Any input sample rate between 15 and 50 kHz can be converted to 44.1 or 48 kHz, selected by a switch on the front panel.

- **Quantization noise imaging**
  The Sample Rate Converter performs Quantization Noise Imaging, which is a form of inband noise shaping which moves the quantization noise outside the audible range. If Q.N.I. is active the output is 16-bit with noise shaping.

- **Scratch Suppression**
  The sensitivity control setting allows an optimal adaption for the source material involved. Because the detection path works on the differential of the input signals, the sensitivity setting may be much lower for a mono recording than, e.g. for a microphone based stereo one. The Sound Enhancer makes it possible to post process a digital recording of a scratched (vinyl) record in order to optimize the parameters.

- **Noise filter**
  Noise reduction is performed by a digital second order low pass filter with an adjustable cut off between 5.5 and 16 kHz.

  At 44.1 kHz: 5.5, 6, 6.5, 7, 8, 9, 10, 11, 12, 13, 14, 15 kHz
  At 48 kHz: 6, 6.5, 7, 8, 9, 10, 11, 12, 13, 14, 15, 16 kHz

  This filter is useful in reducing the surface noise from old gramophone records (78s).

  Compared with an analogue solution this digital implementation of the noise filter ensures that neither extra audible noise nor channel differences in terms of gain, offset, frequency or phase response will occur.

- **Stereo enhancement**
  This function creates a stereo effect from a mono recording. Three elements are involved:
  1. Common phase between left and right channel is avoided.
  2. Room/wall reflections are added, for instance left loudspeaker to left ear. This creates some spaciousness.
  3. Room/wall reflections are added for right loudspeakers to left ear and vice versa.

- **Compression/Expansion**
  The compression/expansion ratio can be adjusted between 10% and 190% in steps of 10% based on 60 dB of the input - dynamic - range. Two examples help to explain this. If the compression ratio is set at 10% and the dynamic range of the source is 60 dB. If the input dynamic range is 90 dB the output dynamic range will be:

  90 dB - 60 dB + 0.1 x 60 dB = 36 dB

  On account of the digital implementation, incorporating delay lines, this function works unobtrusively compared with analogue solutions.

- **Bass/Treble**
  The control range of these filters is ca. +/-10 in steps of 1 dB. The advantages of the digital implementation are applicable in this case as well (see previous paragraph).

- **Spatial Stereo**
  Spatial Stereo widens or narrows the sound stage in three steps. The digitally executed antiphase crossover works without adding any noise or distortion. Default setting: "Normal" Stereo. Minimum setting "Mono".

- **Fader**
  If the fade button is pressed either a "fade in" or "fade out" is triggered. The fade characteristic is in dBs. The fade interval is adjustable between 0.5 and 11 seconds.

- **Volume**
  In case a digital source is selected the maximum volume setting is +6 dB. In this way it is possible to reduce unused headroom of an existing digital recording by a value of up to 6 dB.

5 Sub Code Handling

Note: The Sound Enhancer is transparent for subcodes.

Where the Sound Enhancer is used to interface between a digital source and a digital recorder, for example a DAT player and a CD-R machine, then subcode handling becomes relevant. Most digital audio hardware does not only process audio but also handles additional information needed for,time display, track increment, etc. This information is present in what are known as "subcodes". Each type of digital system has its own type of subcodes, so DAT is different from CD.

The Sound Enhancer will pass on the subcodes received. Thus it is transparent, unlike many sample rate converters. However, because the quality of the various digital recorders in the world is unknown, the manufacturers of the Sound Enhancer cannot guarantee smooth recording in all configurations. In the event that a digital recorder is not designed to handle the various subcodes of the digital sources available certain problems may arise. If only one music track is recorded in most cases no problems will occur. If more related tracks are recorded the time and track increment subcodes information of the source may cause the recorder to malfunction.

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