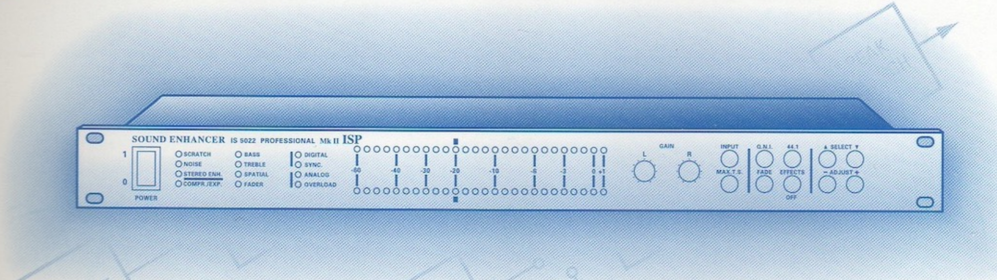
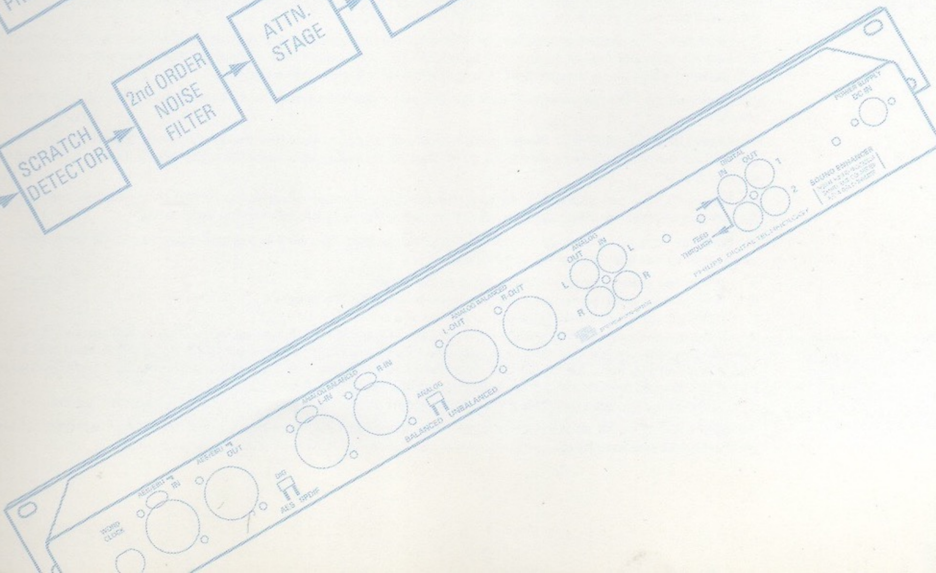
SOUND ENHANCER
IS 5022 PROFESSIONAL

Mark II



THE AUDIO TOOL BOX



INSTRUCTION MANUAL IS 5022 Mark II

WARNING NOTICES

CAUTION

RISK OF ELECTRIC SHOCK
DO NOT OPEN

CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK, DO NOT REMOVE COVER (OR BACK). NO USER-SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.

WARNING: TO PREVENT FIRE OR SHOCK HAZARD, DO NOT EXPOSE THIS EQUIPMENT TO RAIN OR MOISTURE.

This "bolt of lightning" indicates uninsulated material within your unit may cause an electrical shock. For the safety of everyone in your household, please do not remove product covering.

The "exclamation point" calls attention to features for which you should read the enclosed literature closely to prevent operating and maintenance problems.

CAUTION: TO PREVENT ELECTRIC SHOCK, MATCH WIDE BLADE OF PLUG TO WIDE SLOT, FULLY INSERT.

ATTENTION: POUR EVITER LES CHOCS ÉLECTRIQUE, INTRODUIRE LA LAME LA PLUS LARGE DE LA FICHE DANS LA BORNE CORRESPONDANTE DE LA PRISE ET POUSSER JUSQU'AU FOND.

IMPORTANT NOTICE

The Mains adapter MUST be placed such that ventilation is assured, This means that a MINIMUM clearance of 5 cm (2 inches) is available on top of and around the Mains adapter. The Mains adapter must be placed on a hard surface.
Do NOT place the Power Supply Unit on a rug, carpet or any other tissue. Failing to comply with this may result in UNRECOVERABLE thermal shutdown of the Mains adapter.

Important Safety Instructions

1. Read Instructions - All the safety and operating instructions should be read before the appliance is operated.
2. Retain Instructions - The safety and operating instructions should be retained for future reference.
3. Heed Warnings - All warnings on the appliance and in the operating instructions should be adhered to.
4. Follow Instructions - All operating and use instructions should be followed.
5. Water and Moisture - Do not use this appliance near water - for example, near a bath tub, wash bowl, kitchen sink, or laundry tub, in a wet basement, or near a swimming pool, and the like.
6. Carts and Stands - The appliance should be used only with a cart or stand that is recommended by the manufacturer.
7. An appliance and cart combination should be moved with care. Quick stops, excessive force, and uneven surfaces may cause the appliance and cart combination to overturn.
8. Ventilation - Slots and openings in the cabinet are provided for ventilation and to ensure reliable operation of the appliance and to protect it from overheating, and these openings must not be blocked or covered. The openings should never be blocked by placing the appliance on a bed, sofa, rug, or other similar surface. This appliance should never be placed near or over a radiator or heat register. This appliance should not be placed in a built-in installation such as a bookcase or rack unless proper ventilation is provided or the manufacturer's instructions have been adhered to.
9. Grounding or Polarization - This appliance is equipped with a polarized alternating-current line plug (a plug having one blade wider than the other). This plug will fit into the power outlet in only one way. This is a safety feature. If you are unable to insert the plug fully into the outlet, try reversing the plug. If the plug should still fail to fit, contact your electrician to replace your obsolete outlet. Do not defeat the safety purpose of the polarized plug.
10. Power-Cord Protection - Power-Supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them, paying particular attention to cords at plugs, convenience receptacles, and the point where they exit from the appliance.
11. Lightning - For added protection for this product during a lightning storm, or when it is left unattended and unused for long periods of time, unplug it from the wall outlet and disconnect the antenna or cable system. This will prevent damage to the product due to lightning and powerline surges.
12. Overloading - Do not overload wall outlets and extension cords as this can result in a risk of fire or electric shock.
13. Object and Liquid Entry - Never push objects of any kind into this product through openings as they may touch dangerous voltage points or short-out parts that could result in a fire or electric shock. Never spill liquid of any kind on the product.
14. Servicing - Do not attempt to service this product yourself, as opening or removing covers may expose you to dangerous voltage or other hazards. Refer all servicing to qualified service personnel.
15. Damage Requiring Service - Unplug this product from the wall outlet and refer servicing to qualified service personnel under the following conditions:

a. When the power-supply cord or plug is damaged.
b. If liquid has been spilled, or objects have fallen into the product.
c. If the product has been exposed to rain or water.
d. If the product does not operate normally by following the operating instructions. Adjust only those controls that are covered by the operating instructions as an improper adjustment of other controls may result in damage and will often require extensive work by a qualified technician to restore the product to its normal operation.
e. If the product has been dropped or the cabinet has been damaged.
f. When the product exhibits a distinct change in performance - this indicates a need for service.

16. Heat - The product should be situated away from heat sources such as radiators, heat registers, stoves, or other products that produce heat.

IMPORTANT NOTES

INSTRUCTION MANUAL

Instruction Manual - 3104 125 2167.1

The information in this instruction manual is believed to be accurate and reliable and may be changed without notice. No liability will be accepted by the publisher for any consequence of its use. Publication thereof does not convey nor imply any license under patent - or other industrial or intellectual property rights.

Printed in the Netherlands;

Date of first release: 11/94

SCMS

The Sound Enhancer IS 5022 does not comply with the Serial Copy Management System **(SCMS)**.

Recording or copying is only authorised if there is no violation of copyright or other rights of third parties.

WARNING

The Sound Enhancer is delivered with a separate mains adapter. If the mains adapter is connected to the mains and the Sound Enhancer is switched off, there is still a very small current through the transformer of the mains adapter. The mains adapter is designed for that. It is advised, however, in cases of prolonged absence to unplug the mains adapter from the wall socket.

CAUTION: ANY CHANGE OR MODIFICATION TO THE EQUIPMENT BY THE USER NOT EXPRESSLY APPROVED BY THE MANUFACTURER COULD VOID THE USER'S AUTHORITY TO OPERATE SUCH EQUIPMENT.

For U.S.A.

This device complies with part 15 of the FCC (U.S.A.) rules. Operation is subject to the following two conditions:
1. This device may not cause harmful interference, and
2. This device must accept any interference received, including interference that may cause undesired operation.

NOTE: This equipment has been tested and found to comply with the limits for Class A digital devices, pursuant to Part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference in a commercial environment. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

For Canada

This digital apparatus does not exceed the Class A limits for radio noise emissions from digital apparatus as set out in the radio interference regulations of the Canadian department of Communications.

Cet appareil numérique n'émet pas de bruits radioélectroniques dépassant les limites applicables aux appareils numériques de Classe A prescrit dans la règlement sur le brouillage radioélectroniques édicté pas le ministère des communications du Canada.

CONTENTS

For U.S.A.

This device compli[illegible]S.A.) rules. Operation is subject to the following two [illegible]
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2. This device must accept any interfe[illegible]ncluding interference that may cause undesired opera[illegible]

NOTE: This equipment has been tested and found to comply with the limi[illegible] the FCC rules. These limits are designed to provide [illegible] protection against harmful interference in a commercial envir[illegible]his equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instruc[illegible]ause harmful interference to radio communicatio[illegible] of this equipment in a residenti[illegible]rence in which case the user will be require[illegible] at its own expense.

For Canada

This digital appara[illegible] Class A limits for radio noise emissions from digital apparatus as set out in the radio interference regulations of the Canadian department of Communications.

Cet appareil numérique n'émet pa[illegible]dioélectroniques dépassant les limites applicables aux app[illegible]iques de Classe A prescrit dans la réglem[illegible] radioélectroniques édicté pas le ministère des comm[illegible]

1 General Description

1 Introduction

The Sound Enhancer Mark II is a unique combination of:

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

The Sound Enhancer:

- is very easy to use and requires no special skills other than careful listening.
- 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.
- All functions are provided in **REAL TIME**.

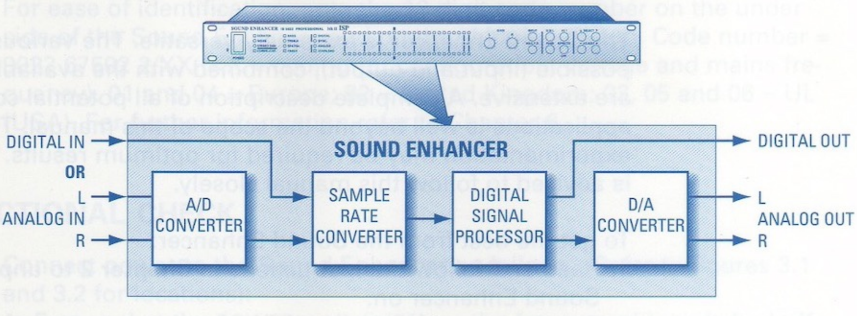


Figure 1.1: Shows an overall functional diagram of the Sound Enhancer.

2 Available functions and effects

The Sound Enhancer provides the following audio functions:

- scratch suppression (de-clicking)
- noise reduction, offering a low-pass filter with an adjustable cut-off between 16 kHz and 5.5 kHz.
- stereo enhancement, allowing a stereo effect to be created from a mono signal
- compression/expansion
- fader (digital)
- 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 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 essentially 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. The box contains:
   - 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 corresponds with the local mains supply.

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, note 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 and 04 = Europe; 02 = United Kingdom; 03, 05 and 06 = 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 main adapter cable with the mini DIN connector to the POWER SUPPLY DC IN (B10) socket on the back panel of the Sound Enhancer.

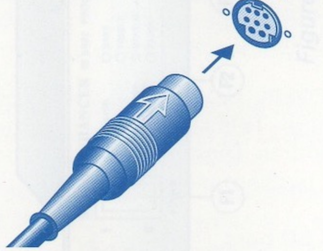


Figure 2.1: Connecting the power adapter.

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.

***Note:** The mains adapter is fitted with a thermal overload protector, which **cannot** be reset once triggered. To avoid unwanted shut off, sufficient cooling must be provided for the mains adapter, refer to important notice printed on the inside cover of this manual.*

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

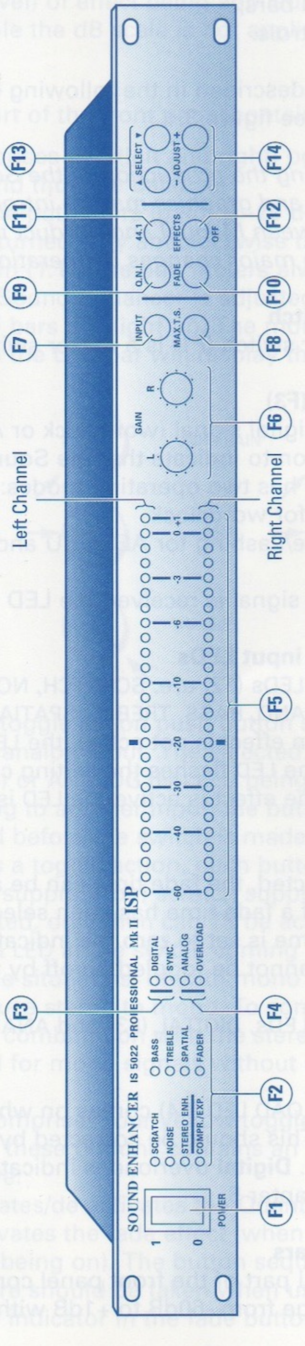


Figure 3.1: Sound Enhancer front panel.

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.

Note: *During the production of the Sound Enhancer small changes in cosmetics and graphics may be introduced. Consequently same differences between Manual and product may occur. These changes do not indicate to major changes in operational characteristics.*

Power switch
The power switch (F1) is a rocker switch which switches D.C. only.

Sync LED (F3)
If a valid digital signal (wordclock or AES/EBU) is received, the SYNC LED will come on to indicate that the Sound Enhancer is synchronised. The SYNC LED has two operating modes:

- steady, for wordclock
- isophase/flashing for AES/EBU and SPDIF.

If no sync. signal is received the LED will not come on.

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

- 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.

Fader (F2)
When selected, the fade time can be adjusted between 0.5 and 11 seconds. If a fade time has been selected, the indicator LED stays on. If the fade time is set to zero the indicator LED will not come on. The fader function cannot be toggled on/off by the EFFECTS button (F12).

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

The OVERLOAD LED (F4) comes on when the **analog** input signal level is too high. This should be corrected by adjusting the the output level of the source. **Digital** overload is indicated by the +1 dB LED of the LED bars; refer to chapter 5.

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.

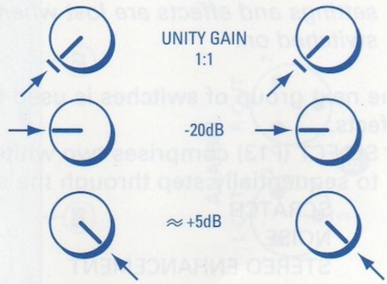
During normal operation, the LED bars display the audio input signal level. The upper row is the left channel, the lower row the right channel.

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 not applicable.

Operator controls

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

- GAIN (F6) comprises two (left and right) potmeters that adjust the level of the audio, and thus the balance.
- The range of the potmeters (gain) is -20 dB to (approx.) +5 dB. If the potmeters are turned fully anticlockwise the Sound Enhancer is adjusted for unity gain (1:1). The pot. meters always operate in the digital domain. If the Sound Enhancer is adjusted for unity gain only the first LED of the LED bars will light up. The moment the potmeters pass the position -20 dB the LED bar will display the gain setting.



- 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 toggle action, push button, switch that improves (TS = transient suppression) scratch suppression. The **scratch** effect must be activated, only then can TS 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. Max. TS can be used for mono signals without any detrimental 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. During fade-out and fade-in the LED indicator in the fade button will flash. (See also F2.)

- 44.1 (F11) sets the output sample rate to 44.1 kHz instead of the default 48 kHz. When the LED in the push button is off, the sample rate is 48 kHz. Pressing the switch a second time reverts the output sample rate back to 48 kHz. In order to minimize noise during switching, the change over is implemented via an automatic "mute". Consequently switching between sample rates is not instantaneous. In order to minimize the risk of accidental switching of the sample rate a delay has been built in. The select button must be pressed for about a second before the switch is made.
- 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 particular effect will be toggled off.

Note: *If the Sound Enhancer is switched off all settings are saved and will be available next time the unit is switched on again. If the Sound Enhancer is switched off while the EFFECT OFF control is active all settings and effects are lost when the Sound Enhancer 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	BASS
NOISE	TREBLE
STEREO ENHANCEMENT	SPATIAL
COMPRESS/EXPAND	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 "+" button increases the effect level and the "-" button decreases it. The effect level selected is indicated by the LED bars.

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

2 Rear panel

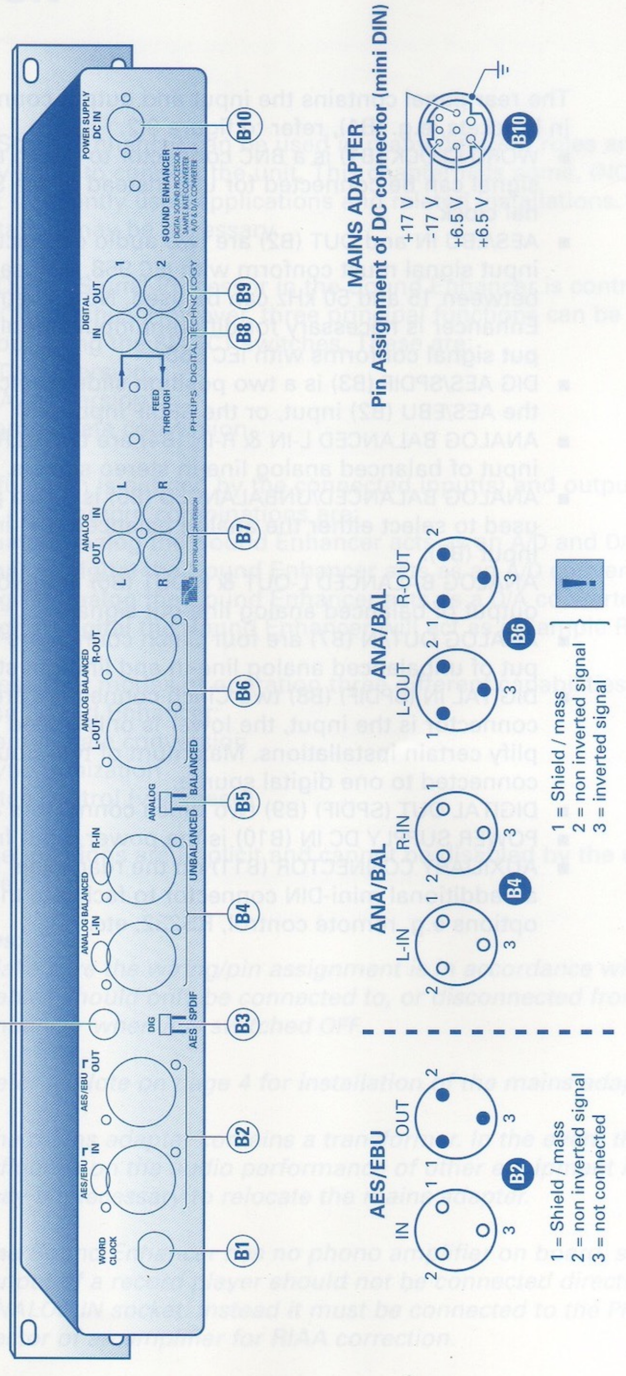


Figure 3.2: Sound Enhancer rear panel and pin assignments.

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

- WORD CLOCK (B1) is a BNC connector to which an external word clock signal can be connected for use instead of the Sound Enhancer's internal clock.
- AES/EBU IN and OUT (B2) are two audio connectors (XLR). 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.
- DIG AES/SPDIF (B3) is a two position slider switch used to select either the AES/EBU (B2) input, or the SPDIF input (B8).
- 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 slider switch used to select either the analog balanced (B4) input, or the unbalanced input (B7).
- ANALOG BALANCED L-OUT & R-OUT (B6) are two XLR connectors for the output of balanced analog line-out 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 (SPDIF) (B8) two Cinch connectors are provided. The top connector is the input, the lower is only a feed through, this may simplify certain installations. Maximum of two Sound Enhancers may be connected to one digital source.
- DIGITAL OUT (SPDIF) (B9) two Cinch connectors are provided.
- POWER SUPPLY DC IN (B10) is the power input from the mains adapter.
- AUXILIARY CONNECTOR (B11) on the rear panel has been prepared for an additional mini-DIN connector to facilitate the installation of future options e.g. remote control, RS232, etc.

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 will act as a Sample Rate Converter.

In these four modes of operation three different capabilities are provided by the Sound Enhancer:

- Sample rate conversion
- Synchronization
- Pitch control (varispeed)

These functions are implicit and cannot be disabled by the user; refer to figure 6.1.

Notes:

1. *Make sure the wiring/pin assignment is in accordance with figure 3.2. Cables should only be connected to, or disconnected from, the Sound Enhancer when it is switched OFF.*

2. *Refer to Note on page 4 for installation of the mains adapter.*

3. *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.*

4. *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.*

2 Sound Enhancer as Sample Rate Converter

The default setting of the output sampling rate of the Sound Enhancer is 48 kHz.

Push button F11 switches the Sound Enhancer to 44.1 kHz. If 44.1 kHz is selected the LED indicator in the push button will light up. The Sound Enhancer will accept:

INPUT	OUTPUT
15 - 50.4 kHz	44.1 kHz
17 - 54 kHz	48 kHz

This range covers most applications:

Sampling Frequency	Application (e.g.)
44.1 kHz	CD, DCC, DAT
32 kHz	NICAM, DSR, DCC, DAT, MAC
48 kHz	DAT, DCC, DAB, S-VHS, prof. audio Eq.
16 kHz	MAC
18.9 kHz	CD-I (level C)
37.8 kHz	CD-I (level A & B)
31.5 kHz	8 mm VCR
44.056 kHz	U-matic VCR
38 kHz	Digital FM stereo decoders

Figure 4.1 Sample rate applications.

4 Installation

3 Synchronization (slave mode)

The Sound Enhancer can act as a slave. In this mode an external clock replaces the internal clock. Two sources of external signal can be handled:

- a signal at the word clock connector
- a valid AES/EBU digital signal at either the XLR or the 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 uses its own internal clock. The SYNC LED is then off.

The Sound Enhancer can be synchronized only for 48 kHz and 44.1 kHz.

The Synchronization modes of the Sound Enhancer are arranged in the following hierarchy:

I wordclock
II AES/EBU or SPDIF
III internal clock, no synchronization.

Note: *The Sound Enhancer is not designed to be a reference clock. When slaving to AES/EBU it may be prudent to disconnect the wordclock cable. When using the external wordclock, connect the coaxial cable direct via a BNC connector, do **NOT** use a "T"-connector with a 75 Ω termination.*

4 Calibration

4.1 Standard calibration

The Sound Enhancer is factory calibrated for an analog input and output signal of 15 dBu (unloaded). This means that when a digital input signal of 0 dB full scale is applied, the balanced analog outputs will be 15 dBu. Furthermore if an analog input signal of 15 dBu is applied, the digital outputs will be 0 dB full scale. This ensures that the full 20 bit digital signal range of the AD converter is used.

If it is required to adjust or to recalibrate the Sound Enhancer first the output must be calibrated then the input can be calibrated. Do not reverse this sequence. The Sound Enhancer can be calibrated for analog input/output signals between 9 and 15 dBu.

4.2 Preparation for calibration

1. Switch the Sound Enhancer off if powered up.
2. Make sure the pot. meters on the front panel are fully turned anti-clock wise, that is: unity gain.
3. Remove cover, use appropriate TORX screwdriver.
4. Locate XLR PCB (refer to figure 4.2).

A digital source is required for calibration of the outputs which can be a CD-player, a DCC or DAT-player etc. It will be more convenient to use a professional test system like the Audio Precision "System One".

4.3. Calibration of the Output

- Locate the trimpotmeters 3108 and 3109
- Select digital input then connect a digital source of 0 dBFs (1 kHz)
- Connect a suitable dB - meter to the analog balanced outputs (L/R)
- Adjust the trimpotmeters so that both outputs give the desired reading. This is always between 9 and 15 dBu.

4.4 Calibration of the Input

- Make sure the analog outputs are calibrated correctly
- Locate the trimpotmeters 3247 and 3248
- Select analog input then apply an analog source of the required level (for 0 dBFs digital) to the balanced inputs (always between 9 and 15 dBu, 1 kHz)
- Connect a suitable analog dB meter to the analog balanced outputs (L/R)
- Adjust the trimpotmeters so that both outputs give the reading that corresponds with the 0 dBFs level as it was defined during the calibration of the outputs. This is always between 9 and 15 dBu.

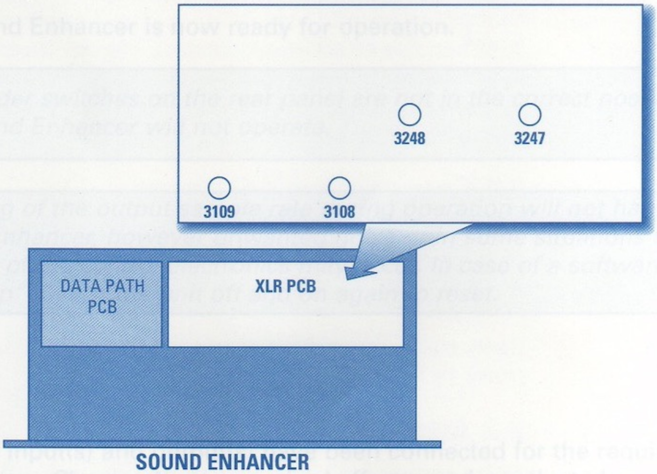


Figure 4.2: Location of trimpots.

4.1 Standard Calibration

[illegible] possible input and output [illegible] where a digital input signal of 0 dB full scale is applied, the balanced analog outputs will be 15 dBu. [illegible] applied, the digital outputs will be 0 dB full scale. This ensures that the full 20 bit digital signal [illegible]

[illegible] Enhancer first the output must be calibrated then the input can be calibrated. Do not reverse this sequence. The Sound Enhancer can be calibrated for analog input/output signals between 9 and 15 dBu.

4.2 Preparation for calibration

1. Switch the Sound Enhancer off if powered up.
2. Make sure the pot. meters on the front panel are fully turned anti-clock wise, that is: unity gain.
3. Remove cover, use appropriate TORX screwdriver.
4. Locate XLR [illegible] (refer to figure 4.2).

A digital source is required for [illegible] the outputs which can be a CD-player, a DCC or DAT-player [illegible] more convenient to use a professional test system like the [illegible] "System One".

4.3. Cali[illegible]

- Locate [illegible]
- Select digital inp[illegible] a digital source of 0 dBFs (1 kHz)
- Connect a suitable dB - meter to the analog balanced outputs (L/R)
- Adjust the trimpotmeters so that both outputs give the desired reading. This is always between 9 and 15 dBu.

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. Make sure that the slider switches B3/B5 on the rear panel are in the required position.
3. Turn the gain control (potmeters) anti-clockwise to the Unity Gain position to avoid excessive volume when the analog input signal is applied.
4. Switch the unit on (POWER 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.
6. Select input (INPUT switch)

The Sound Enhancer is now ready for operation.

NOTE	*If the slider switches on the rear panel are not in the correct position, the Sound Enhancer will not operate.*

Changing of the output sample rate during operation will not harm the Sound Enhancer, however unwanted noise or in some situations malfunction of the control electronics may occur. In case of a software "hang-up" switch the unit off and on again to reset.

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 installed 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
- fade time

(Quantization Noise Imaging refer 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 <u>**will**</u> 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; <u>**or**</u>
- compress/expand.

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

2.2 Activating the effects

This section describes how to activate the different effects.

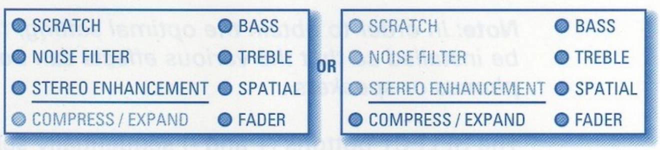


Figure 5.1: Valid combinations of effects

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 boost the scratch suppression capability above the level selected, press the MAX T.S. switch.

If recordings are made from vinyl records, 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 phono cartridge. The scratch suppression is also dependent on the stylus force, too low or high will give less than optimum results. Similarly, lateral pressure (skating) 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 (refer to section 6.4)

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

To activate compression/expansion:
1. Repeatedly press either of the SELECT buttons until the compress/expand LED flashes.
2. Press the ADJUST buttons to obtain the required setting. (If any of the scratch/noise filter/stereo enhancement effects were active, they will be automatically de-activated).
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.

In order to avoid "clipping" when in unity gain mode, automatic scaling is provided via a special anti-clipping algorithm; refer to chapter 6.4.

2.2.4 Spatial

Spatial can widen or narrow the sound stage. The default setting is STEREO, available are 5 steps to mono and 5 steps to full Spatial stereo

Spatial stereo 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.

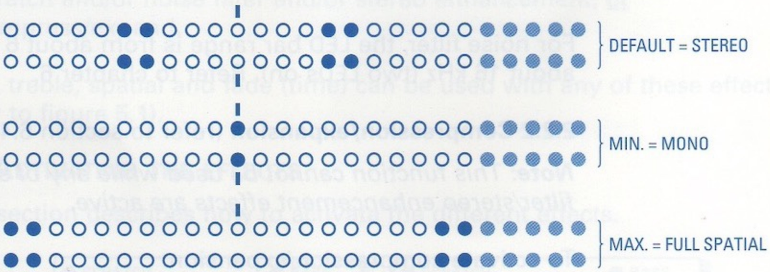


Figure 5.2 LED bar for spatial effect

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. 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 Gain

During normal operation the LED bar shows the audio levels (L and R). The input levels can be individually adjusted via the two GAIN pot. meters. The moment either of these pot meters is adjusted, the LED bar will indicate the setting of the potmeters. After two seconds without adjustment, the display will revert to showing the actual input signal. In the "Unity Gain" position the LED bars will **not** indicate the level.

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 Overload conditions

3.1 Gain Control

The range of the gain control is -20 dB to +5 dB and in addition to this a separate "unity gain" setting is provided. The operator should be aware that the very useful amplification of +5 dB has certain implications which are explained below.

3.2 Analog

When an analog input signal is too high, this situation is indicated by the "OVERLOAD" LED being lit.

3.3 Digital

When using the digital inputs, due to the gain control extending to approximately +5 dB a situation may occur which can be described as "digital overload". The Sound Enhancer follows the standard practise that 0 dB full scale corresponds to the full range of the -60 dB LED up to and including the 0 dB LED of the display.

For 0 dB full scale, clipping may occur if the gain control is set between 0 dB and +5 dB. This is indicated by the red +1 dB LED of the display coming on.

On account of the characteristics of the bass and treble control, (refer to fig. 6.7) hard clipping may occur unless extra attenuation is applied. The Sound Enhancer is fitted with an automatic anti clipping algorithm that attenuates the gain setting to avoid this undesirable situation. The maximum attenuation of this algorithm is 10 dB. This corresponds to the range of the bass and treble control (+ or - 10 steps of about 1 dB). This automatic scaling function is not active on the other effects and functions of the Sound Enhancer.

***Note:** the attenuation does not alter the control span of the potmeters.*

Example: When one increases the bass and/or treble and the GAIN control is set at -10 dB or higher, the automatic attenuation algorithm is activated. For every 1 dB increase of bass/treble (ADJUST +) there is a corresponding overall attenuation of 1 dB when the attenuation algorithm is active.

4 Quantization noise imaging

Quantization Noise Imaging (Q.N.I.) can be used independently 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 have little benefit.

To activate Quantization Noise Imaging:

- 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 disables Q.N.I. If Q.N.I. is inadvertently left on, it will not have any negative influence on the audio.

5 Subcode handling

The sound Enhancer is transparent as far as the "U"-channel information is concerned. The Sound Enhancer complies with IEC 958 for the "C"-channel information. For detailed information on subcode handling refer to section 6.5.

6 General reset

It is possible to execute a general reset of the Sound Enhancer. The general reset procedure can be used to set all the effects/functions to zero or default position, or to cure a "hang-up" which does not respond to switching the unit off and on.

Reset Procedure:

- Switch the Sound Enhancer off
- Simultaneously depress the **44.1** and **SELECT ▼** buttons
- Switch the unit on again, keeping the **44.1** and **▼** buttons depressed for about a second, then release
- Wait for all of the LEDs to switch on
- Press "EFFECTS OFF"
- Rotate the gain potmeters left or right to engage, then set to the required gain level

The Sound Enhancer is now reset and in the default condition with 48 kHz output and digital input selected.

3 Overload conditions

3.1 Gain Control

The range of the gain control is -20 dB to +5 dB and in addition to this a separate "unity gain" setting is provided. The operator should be aware that the very useful amplification of +5 dB has certain implications which are explained below.

3.2 Analog

When using [illegible] is indicated by the [illegible]

3.3 Digital

When using the digital inputs, due to the gain control extending to approximately +5 dB a situation may occur which can be described as "digital overload". The Sound Enhancer follows the standard practise that 0 dB full scale corresponds to the full range of the -60 dB LED up to and including the 0 dB LED of the display.

For 0 dB full scale, clipping may occur if the gain control is set between 0 dB and +5 dB. This is indicated by the red +1 dB LED of the display coming on.

On account of the characteristics of the bass and treble control, (refer to fig. 6.[illegible]) hard clipping may occur unless extra attenuation is applied. The Sound Enhancer is fitted with an automatic anti clipping algorithm that attenuates the gain setting to avoid this undesirable situation. The maximum attenuation of this algorithm is 10 dB. This corresponds to the range of the bass and treble control (+ or - 10 steps of about 1 dB). This automatic scaling function is not active on the other effects functions of the Sound Enhancer.

Note: the attenuation does not alter the control span of the potmeters.

Example: When one increases the bass and/or treble and the GAIN control is set at -10 dB or higher, the automatic attenuation algorithm is activated. For every 1 dB increase of bass/treble (ADJUST +) there is a corresponding overall attenuation of 1 dB when the attenuation algorithm is active.

6 Technical Data

Unless otherwise indicated, all values are nominal or typical.

1 General

Weight:

Sound Enhancer:	4.3 kg
Mains Adapter:	1.1 kg

Dimensions: mm

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

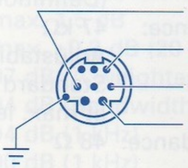
Power:

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

Type:	SBC 5021/01 Europe, /02 UK, /03 USA (UL) ISP marked products
Mains voltage:	230 V ±15% or 120 V ±15%
Mains frequency:	50 Hz, or 60 Hz
Consumption:	approx. 25 W

MAINS ADAPTER

Pin Assignment of DC connector (mini DIN)



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
Analog:	Line-out stereo (L/R) Connector: Cinch/RCA Balanced out (L/R), Connector: XLR

2 Performance

DIGITAL

20 bits audio-performance

Balanced:	XLR connectors
Unbalanced:	Cinch connectors
Input frequency:	17-54 kHz/15-50.4 kHz
Output frequency:	48 kHz/44.1 kHz

Word clock

Input impedance:	45 kΩ/BNC connector
Input level:	min. 2.6 Vpp, max 6 Vpp
Frequency:	48 kHz or 44.1 kHz (tolerance: ±50 ppm)

ANALOG

Balanced (XLR)

Input sensitivity:	Adjustable: +9 dBu to +15 dBu Standard factory adjustment: +15 dBu ±0.5 dB for max. level (0 dBFs digital full scale) (Definition: 0 dBu = 0.775 Vrms; 1 mW in 600 Ω)
Input Impedance:	47 kΩ
Output level:	Adjustable: +9 dBu to +15 dBu Standard factory adjustment: +15 dBu ±0.5 dB at max. level (0 dB digital full scale)
Output Impedance:	48 Ω

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

Unbalanced (Cinch/RCA)

Input sensitivity:	0.5 V rms for maximum level (0 dB dig) -1 dBV for 0 dBFs (0 dBV = 1 Vrms) at unity gain: 0 dB **2 Vrms +2 dB (2.5 Vrms) maximum allowable**
Input Impedance:	50 kΩ
Output level:	2 V rms for maximum level (0 dB digital)
Output Impedance:	200 Ω

Balanced (XLR)

Analog in to analog out (complete analog path)

Unbalance:	max. 2 dB
Amplitude linearity:	max. ±0.2 dB (20 Hz - 20 kHz)
S/N ratio:	96 dB (A weighted) 94 dB (Bandwidth 20 kHz)
Dynamic Range:	96 dB (1 kHz)
THD + Noise:	90 dB (1 kHz)
Channel separation:	100 dB (1 kHz)

Analog In

Unbalance:	max. 1 dB (adjusted)
Amplitude linearity:	max. ±0.1 dB (20 kHz - 20 kHz)
S/N ratio:	101 dB (A weighted)
	97 dB (Bandwidth 20 kHz)
Dynamic Range:	97 dB (1 kHz)
THD + Noise:	91 dB (1 kHz)
Channel separation:	100 dB (1 kHz)

Analog Out

Output Voltage:	+15 dBu ±0.5 dB
Unbalance:	max. 1 dB (adjusted ±0.05 dB)
Amplitude linearity:	max. ±0.1 dB (20 Hz - 20 kHz)
S/N ratio:	100 dB (A weighted)
	95 dB (Bandwidth 20 kHz)
Dynamic range:	95 dB (1 kHz)
THD + Noise:	95 dB (1 kHz)
Channel separation:	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); input: -1 dBV, unity gain

Unbalance:	max. 1.5 dB
Amplitude linearity:	max. ±0.2 dB (20 Hz - 20 kHz)
S/N ratio:	97 dB (A weighted)
	94 dB (Bandwidth 20 kHz)
Dynamic range:	94 dB (1 kHz)
THD + Noise:	90 dB (1 kHz)
Channel separation:	100 dB (1 kHz)

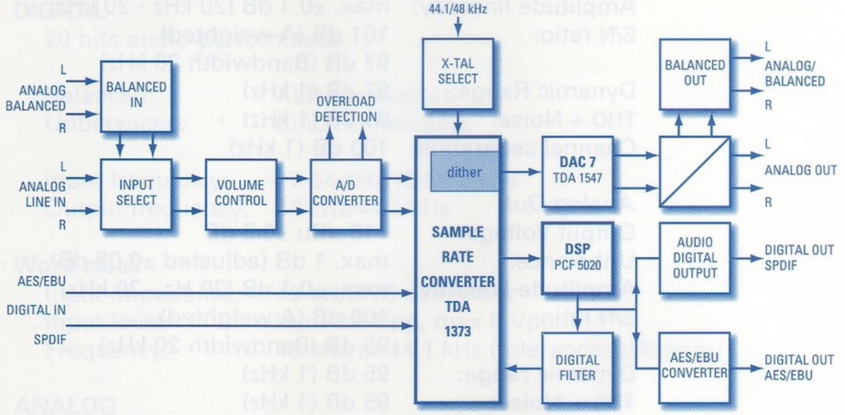
Analog In; input: -1 dBV, unity gain

Unbalance:	max. 0.8 dB
Amplitude linearity:	max ±0.1 dB (20 Hz - 20 kHz)
S/N ratio:	103 dB (A weighted)
	98 dB (Bandwidth 20 kHz)
Dynamic range:	98 dB (1 kHz)
THD + Noise:	92 dB (1 kHz)
Channel separation:	105 dB (1 kHz)

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)
S/N ratio:	103 dB (A weighted)
	98 dB (Bandwidth 20 kHz)
Dynamic range:	95 dB (1 kHz)
THD + Noise:	95 dB (1 kHz)
Channel separation:	110 dB (1 kHz)
Low level linearity:	within 1.5 dB at -90 dB

3 KEY COMPONENTS



A/D Converter	AD MOD 79
D/A Converter	TDA 1547
DSP function	PCF 5020D
ADOC function	CS 8401
ADIC function	TDA 1373
SRC function	TDA 1373

NOISE SHAPER function with **dither**

Figure 6.1: Sound Enhancer block diagram.

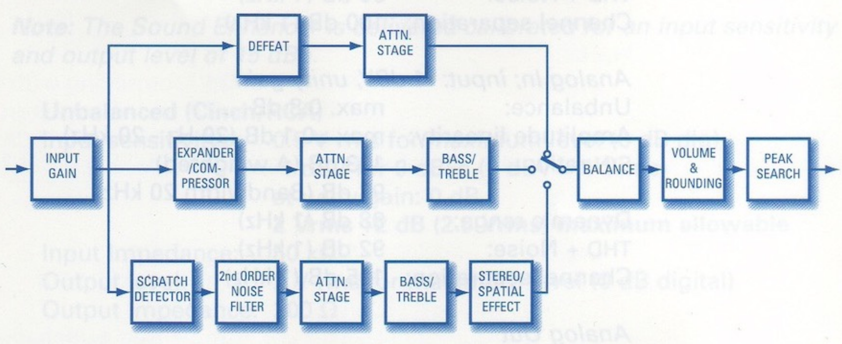


Figure 6.2: Sound Enhancer DSP block diagram.

4 Functional data

- **Defeat mode**
  If no effects are applied (defeat mode), the Sound Enhancer will cause a minor time lag of between 1 and 3 ms.

- **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 48 or 44.1 kHz.

- **Quantization noise imaging**
  Quantization Noise Imaging 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 this particular form of in-band noise shaping. Although the Sound Enhancer operates at 24-bits internally and the A/D conversion is 20 bits this does not automatically imply 20-bit resolution (refer to figure 6.3).

IN	Q.N.I.	OUT
Digital 20-bit	-	**Digital** 20-bit
Digital 20-bit	Q.N.I.	**Digital** 16-bit resolution + noise shaping
Digital 20-bit	-	**Analog** 16-bit resolution
Analog	-	**Digital** 20-bit
Analog	Q.N.I.	**Digital** 16-bit resolution + noise shaping
Analog	Q.N.I.	**Analog** 16-bit resolution + noise shaping
What is applicable for 20-bit input signals also applies for 18-bit input signals		

Figure 6.3 Q.N.I. effect on output resolution.

- **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. The de-click function generates a small time delay of about 10 ms.

- **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 Sound Enhancer is fitted with a fixed set of 10 expansion and compression curves. The curves are presented in figures 6.4 and 6.5. The normal (stereo) situation is presented by the straight line in each diagram. It is possible either to expand or to compress the audio by a factor of two (maximum setting), as the diagrams show.

  For instance: COMPRESSION
  a Δ of 10 dB at the input will result in
  a Δ of 5 dB at the output.

  EXPANSION
  a Δ of 10 dB at the input will result in
  a Δ of 20 dB at the output

  Each step on the LED display corresponds with one curve, the amount of expansion or compression as a percentage of the input signal is shown in figure 6.6.

6 Technical Data

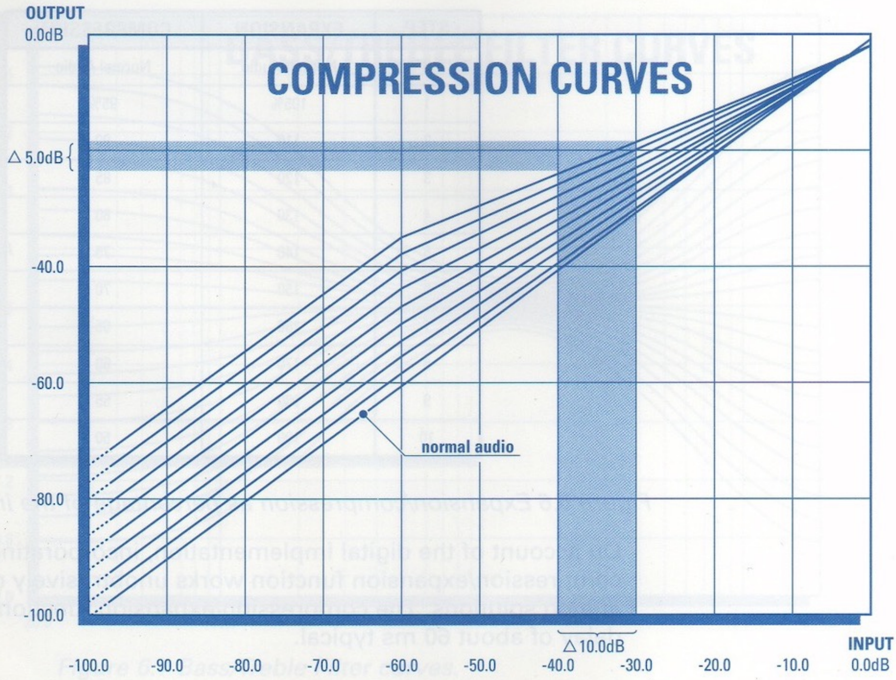


Figure 6.4 Compression curves.

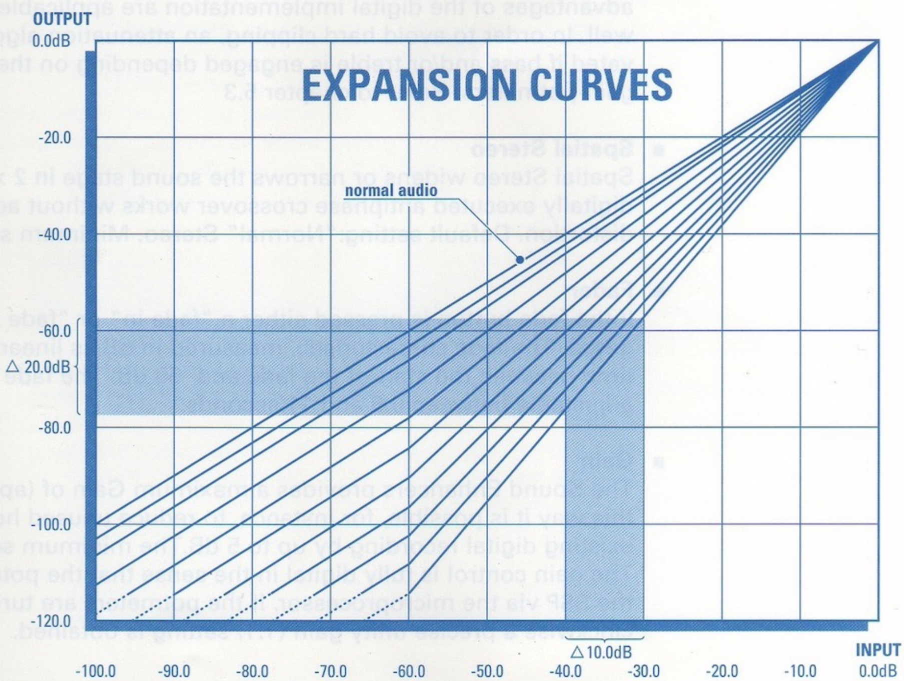


Figure 6.5 Expansion curves.

STEP	EXPANSION	COMPRESSION
0	Normal Audio	Normal Audio
1	105%	95%
2	110	90
3	120	85
4	130	80
5	140	75
6	150	70
7	160	65
8	170	60
9	180	55
10	190	50

Figure 6.6 Expansion/compression as percentage of the input level.

On account of the digital implementation, incorporating delay lines, the compression/expansion function works unobtrusively compared with analog solutions. The compression/expansion function does cause a delay of about 60 ms typical.

- **Bass/Treble;** (Refer to figure 6.7)
  The control range of these filters is in ±10 steps of about 1 dB. The advantages of the digital implementation are applicable in this case as well. In order to avoid hard clipping, an attenuation algorithm is activated if bass and/or treble is engaged depending on the position of the gain potmeters. Refer to chapter 5.3

- **Spatial Stereo**
  Spatial Stereo widens or narrows the sound stage in 2 x 5 steps. The digitally executed antiphase crossover works without adding 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, measured in dB, is linear as a function of time between the start of the fade and -50 dB. The fade interval is adjustable between 0.5 and 11 seconds.

- **Gain**
  The Sound Enhancers provides a maximum Gain of (approx.) +5 dB. In this way it is possible, for instance, to reduce unused headroom of an existing digital recording by up to 5 dB.The minimum setting is -20 dB. The gain control is fully digital in the sense that the potmeters control the DSP via the microprocessor. If the potmeters are turned fully anticlockwise a precise unity gain (1:1) setting is obtained.

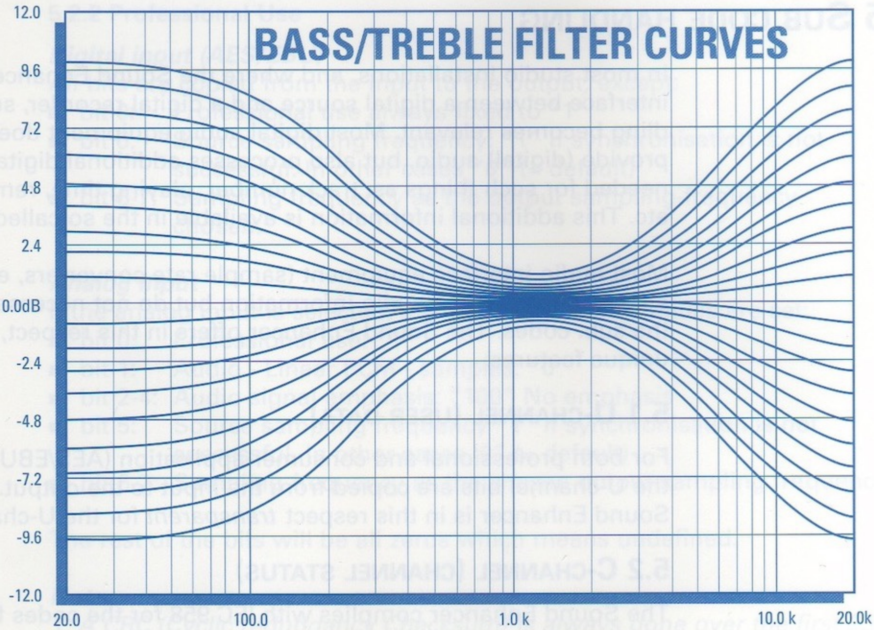


Figure 6.7 Bass/Treble Filter curves.

5 Sub code handling

In most studio installations, and where the Sound Enhancer is used to interface between a digital source and a digital recorder, subcode handling becomes relevant. Most digital audio equipment does not solely provide (digital) audio, but also processes additional digital information needed for such things as: track number, playing time, remaining time etc. This additional information is available in the so-called sub. codes.

Most audio interface equipment (sample rate converters, effect boxes etc.) will pass on the audio information but do not necessarily pass on the sub. codes. The Sound Enhancer offers in this respect, some rather unique features:

5.1 U-channel (user data)

For both professional and consumer application (AES/EBU and SPDIF), the U-channel bits are copied from the input to the output. Thus the Sound Enhancer is in this respect *transparent* for the U-channel.

5.2 C-channel (channel status)

The Sound Enhancer complies with IEC 958 for the codes for the C-channel, with the following exceptions:

5.2.1 Consumer use (Cinch/RCA)

Digital input (SPDIF)

All bits are copied from the input to the output, except:

- bit 0: Consumer use always fixed to “0”
- bit 2: Copy bit: always fixed to “1”: no copyrights asserted
- bit 24-27: Sampling frequency as the output sampling-frequency chosen.

Analog input

If the analog input is selected, the next default values are set:

- bit 0: Consumer use: “0”
- bit 1: Audio - Linear PCM - samples: “0”
- bit 2: Copy - bit: “1”: no copyrights asserted
- bit 3-5: Audio signal emphasis: “000” No emphasis
- bit 8-15: Category Code General: all zeros
- bit 24-27: Sample frequency as chose (44.1/48 kHz)
- bit 28-29: Clock accuracy: “10”: Level I

5.2.2 Professional Use

Digital input (AES/EBU)

All bits are copied from the input to the output, except:

- bit 0: Professional use always fixed to "1"
- bit 5: Source sampling frequency: "1" if synchronisation is not successful. In other cases "0" (= default)
- bit 6-7: Sampling frequency as the output sampling-frequency chosen.

Analog input

If the analog input is selected, the following default values are set:

- bit 0: Professional use: "1"
- bit 1: Audio - Linear PCM - samples: "0"
- bit 2-4: Audio signal emphasis: "100" No emphasis
- bit 5: Source sampling frequency "1" if synchronisation is not successful. In other cases '0" (= default)
- bit 6-7: Sampling frequency as the chosen output sampling frequency

The rest of the bits will be all zeros which means undefined.

Note:

1. A CRC (Cyclic redundancy Checksum) is always done over the first 23 bytes (byte 0-22) and the CRC - character is put in byte 23.

2. The treatment of Bit-5 warrants special attention, since not all audio equipment complies with the IEC 958. To summarize the Sound Enchancer will set Bit-5 to "0" in case of external synchronization. In case synchronization is not successful the value of Bit-5 will be set to "1". The default setting of Bit-5 is "0" also if synchtonization is not applicable.

3. Also the treatment of Bit-2 in SPDIF ("copy protect") warrants extra attention. Bit-2 is always set to "1", which means: ***copy permitted.***

SCMS
The Sound Enhancer IS 5022 is intended for professional use it therefore does not comply with the serial copy management system.

Great care was taken to provide practical and reliable subcode handling. Because the quality of the various digital recorders and sources in the world is unknown, the manufacturers of the Sound Enhancer cannot guarantee smooth operation in all situations.

In the event that a recorder is not designed to handle the appropriate subcodes of the digital sources available, problems may arise. If only one music track is recorded usually no problems will occur. If more tracks are recorded and the time and track increment information is not properly processed malfunctioning of the recorder may occur.

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