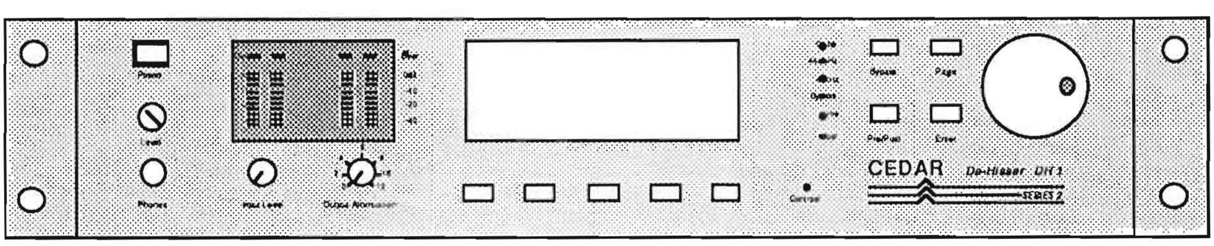
CEDAR

Professional Hardware Systems

DH-1 De-Hisser

Digital Audio Restoration System

SERIES 2



OWNER'S MANUAL

TABLE OF CONTENTS

INTRODUCTION

Thank you for purchasing the CEDAR DH-1 De-Hisser Module. This is the world's most advanced dedicated single-ended noise removal unit, and offers processing power and performance that could only previously be obtained using digital signal processors (DSPs) installed in desk-top (or larger) computer systems such as the CEDAR Production System. The De-Hisser is designed for professional use, although it will work perfectly well in a domestic environment, and its features include the following:

- Revolutionary noise removal algorithms
- No need for a “Spectral Fingerprint”
- The latest 'SERIES-2' CEDAR hardware
- Digital Audio interfaces conforming to the AES/EBU and SP-DIF standards
- 24-bit input and output resolution when using AES/EBU interfaces
- Three sample rates supported on digital inputs: 32kHz, 44.1kHz and 48kHz
- Two sample rates supported on analogue inputs: 44.1kHz and 48kHz
- Balanced analogue inputs and outputs for connection to professional analogue equipment
- ADC and DAC converters using the latest 64x over-sampling Δ-Σ (Delta-Sigma) technology
- >103dB dynamic range A/D and >93dB dynamic range D/A
- Mountable in a 19" EIA rack
- Remote control via MIDI and RS232 interfaces
- SMPTE/EBU timecode capabilities via optional upgrade
- Input and output LED bar-graph VU meters
- Twin 40-bit floating point DSP processors delivering 50MFlops to handle the most complex audio processing requirements
- High levels of artificial intelligence designed into the DH-1 program algorithms making it extremely simple to use

THE BACKGROUND TO CEDAR NOISE REMOVAL

Cheap digital audio (i.e. CD) has made discerning listeners quite intolerant of the noises and distortions present in analogue audio signals. After all, in a perfect digital world there are no clicks, crackle, pops, buzzes or hums, and no hiss - so it's a shame that we live in a far from perfect world. Even today, the vast majority of mixing desks still have all-analogue signal paths, so most DDD-classified CDs are still mastered through numerous analogue stages. And 'vintage' (i.e. pre-1982) recordings are by definition re-mastered from analogue master tapes which inevitably suffer from at least one of the degradations listed above. So recording engineers are turning more and more to the technologies available for reducing any noise added in the signal path, or for removing it from the final recording.

You can rid yourself of any broadband noise you care to mention... white noise, tape hiss, microphone noise, rumbles... you name it, you can eliminate it. Totally, and without any fuss or expensive equipment. How? Simple... by turning your master volume control to zero. OK, so this method also has a rather drastic effect on the signal content of your recording - it completely removes it - but what do you mean, you want to get rid of all the noise but keep the genuine signal absolutely untouched?

Before proceeding any further, perhaps it would be best to describe what we mean by the term 'broadband' noise. Such noise is, by definition, a random effect which adds (or subtracts) a random amplitude at all times to (or from) all frequencies within the audio spectrum. Thus, the term is used inappropriately to describe artefacts such as intermittent electrical clicks or microphone 'grounding'. These problems produce clearly identifiable events of limited duration, and may be corrected by the CEDAR DC-1 De-Clicker and CR-1 De-Crackler using quite different methods to those described below. On the other hand, broadband noise is constantly present (to a greater or lesser degree) in every signal. It is often most intrusive at high frequencies, where the masking effect of loud sounds is least present, so the term 'hiss' is often used to describe all forms of broadband noise.

Firstly, let's dispel any illusions regarding the Dolby B, Dolby C, and dbx noise reduction systems. These are dual-ended processes designed to minimise the accumulation of any extra noise added by the limitations of analogue recording tape. (Dual-ended processes are commonly called encode/decode systems because the recording process 'encodes', and the playback process 'decodes', the signal.) Neither the Dolby processes nor dbx enable you to remove noise from within a signal that already contains it - they simply stop you adding too much more when you commit that signal to tape and then play it back again. Perversely then, both Dolby and dbx help your tape deck to accurately record, and then faithfully reproduce, any noise contained in the original signal. So, what you need is a 'single-ended' process that can remove noise from your signals prior to committing them to tape, or at the very least, can improve the signal to noise (S/N) ratio without affecting the signal adversely. Which brings us neatly back to the volume control... stunningly effective at removing noise, it does nothing to improve the S/N ratio, and has an all-too-noticeable side-effect. No noise, No signal.

The first stage in our evolutionary tale of noise removal is the simple treble filter (or 'low-pass filter'). Less damaging than the volume control which removes the signal altogether, the treble filter only removes a proportion of any signal present above a given frequency (known as the shelf frequency of the filter). Unfortunately, if, at the given frequency, you reduce the amplitude of the noise content of your recording by 6dB (thus making the noise half as loud) you will also reduce the genuine signal at this frequency by the same amount. This will be fine if your recording has little or no high frequency content, but natural sounds and modern electronic instruments have frequency responses up to and beyond the limits of human hearing. Consequently, the treble filter will only be successful in processing your antique collection of '78's, and even then only at a cost.

But this gives us a hint as to how a more effective single-ended noise reduction system could be designed: perhaps a device could be built which removes the high frequencies when there is no signal present, but leaves them untouched when the noise is being masked by genuine high frequencies? Of course it can. It's a Dynamic Filter (so called because the shelf frequency of the filter moves dynamically up and down the frequency spectrum according to information contained in the signal). But such devices are limited: for one thing, they can only remove the noise which exists above the highest frequency of the music present at any given moment. Secondly, they are based on filters with roll-offs typically of the order -12dB/octave or -6dB/octave, so they always allow some high frequencies through, even when they think that they're removing it. And thirdly, even though the filters are designed to track the signal very quickly, they cannot respond instantaneously, so they tend to round off fast transients such as snare drums and samples (which have a habit of introducing high frequencies very rapidly into the signal). And, because their raison d'être is to reduce the signal bandwidth they also tend to dull the genuine signal quite perceptibly. So to summarise dynamic filters: if you're not compromising the signal you may not be removing as much noise as you wish, and if you're removing all the noise you're probably damaging the genuine signal.

Perhaps an alternative approach could give better results? Instead of altering the frequency response of the signal to reduce the noise content, how about changing the volume (amplitude response) of the signal in some way? This isn't such a strange idea. Consider: if noise is always present in a signal then, if the total signal amplitude drops down to the noise level, surely all the genuine musical signal has disappeared? While there are many flaws in this argument (largely to do with the statistical nature of broadband noise) it is, as a generalisation, nearly true. This then suggests a device which will eliminate some of the noise: a Noise Gate. This simple device detects when the signal drops below a certain level (a 'threshold' set by the user) and then cuts off the signal entirely. It's just like turning the volume control of your hi-fi to zero between tracks, and then turning it back up at exactly the moment the music starts again. There are many enhancements to the Gate idea, such as variable attack and release times, and hysteresis (all added to limit the occurrence of damaging side-effects) but the principle always remains the same: if the device decides that there is only noise present at its input, it totally shuts off the signal. There is an adage that says that a multi-track studio can never have too many noise gates because, while they are pretty useless at stereo mastering, they are invaluable for shutting off the intrusive hisses, hums, and buzzes of temporarily unused synths and guitar channels.

Unfortunately, though the Noise Gate sounds great in theory, it doesn't sound so great in reality. It has a distinct advantage over the low-pass filter (after all, it removes all the frequencies of broadband noise, not just the high ones) but once the gate is 'open' all the noise comes flooding back. And if you adjust the threshold so that noise can only come through when the signal is loud enough to mask it, you'll lose the ability to include quiet passages in your recordings. So the Noise Gate is as damaging its own way as the filter is in its. Fortunately, just as the filter can be improved by making it dynamic, so can the gate. Such a device is called an Expander and its operation is a bit like that of a compressor, but in reverse. The Expander still has a threshold control, but unlike the gate (which shuts the volume down to zero once the threshold is passed) the Expander applies a progressive gain reduction, the amount of which is determined by the settings selected by the user. For example, if a signal drops 3dB below the threshold, the Expander may reduce the signal volume by 6dB, 12dB, or any other figure, depending upon the expansion ratio requested. Unfortunately, many audio professionals claim that the subjective difference between the true noise gate and the expander are very small - and you wouldn't use either for top quality recording or mastering.

Some of the more highly specified noise reduction units now feature a combination of dynamic filtering, expansion, and even compression and excitation - effects which have been included to overcome some of the undesirable side-effects of the noise reduction processes. But they are only partially successful when cleaning up complete mixes, and you still can't master full bandwidth CDs or film soundtracks with them. The results simply aren't good enough.

Now, consider what happens when the single-band expander described above encounters a very quiet signal... it further reduces the volume. But what if there is still a significant signal at (say) 3kHz, but very little elsewhere in the frequency spectrum? The single band expander has no way of knowing that there's an important genuine signal within a limited range, and it shuts this out at the same time as all other frequencies. What's needed is an expander which can detect such instances, and cope with them appropriately. A multi-band unit does this by separating the audio spectrum into a number of bands, and then treating each of these as individual signals. Consequently, such a device can be reducing the volume in one band, while passing the signal untouched in other bands. This type of noise reduction has now found its way from rackmount modules onto computers. Digital Audio Workstations (usually hard disk editors with other functions added) utilise processor chips known as Digital Signal Processors (DSPs) which perform millions of calculations every second upon the audio data produced by CD players, DAT machines, and Analogue-to-Digital Converters. Splitting the audio spectrum up into multiple bands is simple for such devices, and applying expansion in the digital domain is a straightforward process compared to some of the more esoteric DSP functions. But even multi-band units have no way to make a true distinction between signal and noise. They still act upon the mistaken assumption that, if the signal level approaches its noise floor, all that is present is broadband noise. Consequently, even the most sophisticated downwards expanders and dynamic filters inevitably remove some of the genuine signal. The consequences of this are well understood and, to a greater or lesser extent, unavoidable: loss of high frequencies, loss of ambience, and degradation of hard transients.

So we finally arrive at the most sophisticated noise removal technology yet implemented: Spectral Subtraction. But to explain what this is we must first dive into a little simple mathematics...

All the methods and products described above use filters, gain controls, or a combination of both to achieve their results. Whether implemented in the analogue or digital domains, all such filters and gain controls are 'ratio' devices - that is, if (at any given frequency) you remove half the power of the noise, you remove half the power of the signal at that same frequency; if you remove 3/4 of the noise, you remove 3/4 of the signal... and so on. But now let's consider what else is possible in the digital domain: Imagine a signal that has, at a given frequency, 100 units of 'volume' on some arbitrary scale. Let's also say that, by measuring the noise content of the signal during an otherwise silent moment, you have determined that there are 20 units of noise present on the same scale. It should be possible to remove the noise amplitude by subtracting these 20 units (in the digital domain) or by filtering out 20% of the signal in the analogue domain. But what if, a moment later, the 'volume' of the signal drops to 40 units? The analogue filter, set to 20% reduction, will only remove 8 units of noise, whereas the digital process is still able to remove the full 20 units (equivalent to a filter reduction of 50%). This is, of course, what we want, because the noise now represents, in fact, 50% of the total signal amplitude. No analogue device can precisely emulate this 'subtractive' filter, and herein lies the power of the computerised noise reduction system.

Computers can, among other things, split the audio signal into hundreds of very narrow bands, and apply Spectral Subtraction to each of these. Splitting the signal this way means that you can be very precise about how much noise you remove, subtracting a lot at (say) 8kHz, while leaving 8.1kHz virtually untouched. Sounds too good to be true? Unfortunately, it is. The noise spectrum of a recording (the spectral fingerprint) can only be accurately measured if there is an otherwise silent passage within the music. If the fingerprint is wrong (maybe because you have captured some lingering reverb, or because a compressor has been applied at some time, or because the original recording engineer has faded sections in and out of the recording) the amount subtracted will be wrong, leading to some very unpleasant sounding side-effects. And, just to make matters worse, many tracks are 'close edited' - the run-in and run-out of the track have been removed - making it impossible to take a fingerprint.

Let's assume that you have a perfect fingerprint. You might expect to produce a very good restoration of your track: large amounts of noise removed, with little or no side-effects. Yet experience shows that all attempts to use an unmodified noise fingerprint lead to a dry and dull sounding result. This is because the fingerprint is merely a snapshot of the noise content of the material, accurate only at the instant at which it is taken. The very essence of noise is its random nature, and because the profile of the noise content is constantly changing, it is necessary for the noise fingerprint within the system to change as well. Which brings us to CEDAR HISS-2. With a noise fingerprint that is updated 44 times per second (allowing CEDAR to track variations in the noise content of the recording); algorithms which 'look-ahead' at the incoming signal, responding to transients before they occur; and an ambience control which ensures that sounds are not prematurely cut short, CEDAR combines many of the analogue and digital ideas discussed above. This means that, in theory at least, the amount of noise being removed is always appropriate. Consequently, HISS-2 avoids the pitfalls and finally enables the user to remove the right amount of noise without damaging the source signal.

But HISS-2 cannot be implemented in a stand-alone box such as the DH-1. It still requires a noise fingerprint, whether captured from the signal or created by the CEDAR operator. (This requirement is almost the sole reason for the 5 years of further research that has occurred since the launch of the original CEDAR Noise Reduction System and the DH-1.) The DH-1 includes revolutionary new algorithms which have finally dispensed with this requirement, enabling you to remove noise in a powerful, yet automated, fashion. The DH-1 will itself analyse the noise content of a signal (whether genuine sound is present or not!) and apply all the power of CEDAR's latest noise removal algorithms to this signal.

The results speak for themselves.

SAFETY INSTRUCTIONS

CAUTION:

1. **Read all of these instructions**
   All safety and operating instructions should be read before the DH-1 is operated.

2. **Save these instructions for future reference.**

3. **Follow all warnings and instructions.**

4. **Water and Moisture**
   The DH-1 should not be used near water, and must not be exposed to rain or moisture. If the DH-1 is brought directly from a cold environment into a warm one, moisture may condense inside the unit. This, in itself, will not damage the DH-1, but may cause hazardous electrical shorting to occur. This could severely damage the DH-1, and even cause danger to life. ALWAYS allow time for the DH-1 to naturally reach ambient temperatures before connecting the mains power.

5. **Mounting**
   The DH-1 should be carefully mounted in a 19" EIA rack, or placed on a flat, stable surface. If used on a cart or free stand, care should be taken when moved: uneven surfaces or excessive force may cause cart and DH-1 to overturn. Do not position the DH-1 in a place subject to strong sunlight, excessive dust, mechanical vibration or periodic shocks.

6. **Wall or Ceiling Mounting**
   The DH-1 has not been designed for mounting directly to walls or ceilings.

7. **Ventilation**
   Good air circulation is essential to prevent internal heat build-up within the DH-1. The DH-1 should be situated so that its position does not interfere with proper ventilation. The DH-1 should not be placed in any situation which impedes the flow of air through the vents at the front and rear. Do not place the DH-1 on a soft surface.

8. **External Heat Sources**
   The DH-1 should be installed away from significant heat sources such as radiators, and (if possible) away from other audio devices such as amplifiers that produce large amounts of heat. Installation in racks with devices such as signal processors or tape machines should not be a problem.

9. **Power Sources**

   The DH-1 features an auto-switching power supply which will work safely on any mains supply in the ranges 95v/130v and 190v/260v, 50Hz or 60Hz AC only.

   You should never attempt to modify or adjust the internal power supply in any way. It contains no user serviceable parts.

10. **Grounding or Polarisation**
The DH-1 should always be grounded (or 'earthed').

11. **Power Cord Protection**
Power connectors should be routed so that they will not be walked on or pinched.

12. **Extended Periods of Non-Use**
The DH-1 is not disconnected from the mains power as long as it is connected to the wall outlet, even if the unit itself has been switched off. Therefore, if the DH-1 is not to be used for an extended period of time, unplug the unit from the wall. Pull the connector out by the plug, never by the cord itself.

13. **Cleaning**
Clean only with a dry cloth. NEVER use liquid cleaners such as alcohol or benzene on the DH-1. NEVER use abrasive pads on the DH-1.

14. **Damage Requiring Service**
The DH-1 should be returned to qualified service personnel when:
    - objects have fallen into the unit
    - liquid has been spilled into the unit
    - the unit has been exposed to rain
    - the unit fails to function or appears to operate abnormally
    - the unit has been dropped, or the case damaged.

15. **Servicing**
The user should not attempt to service the DH-1 beyond the instructions contained in the User's Manual. All other servicing should be referred to qualified service personnel.

SET UP

1. UNPACKING AND INSPECTION

Be careful not to damage the DH-1 during unpacking. Save the carton and all packing materials since you may need them to transport the DH-1 in the future.

In addition to the packaging, the carton should contain the following:

- mains connection lead
- this manual
- blanking plates which may be used to replace the rack-mount ears
- DH-1 Tutorial DAT

2. INSTALLATION SITE

The DH-1 may be used in most areas, but to maintain reliability and prolong operating life observe the following environmental considerations:

- Nominal temperature should be maintained between 5° and 35° Centigrade (41° and 95° Fahrenheit).

- Relative humidity should be in the range 30% to 60% non-condensing.

- Strong magnetic fields should not exist nearby.

3. RACK MOUNTING

The DH-1 can be mounted in a standard 19" EIA rack.

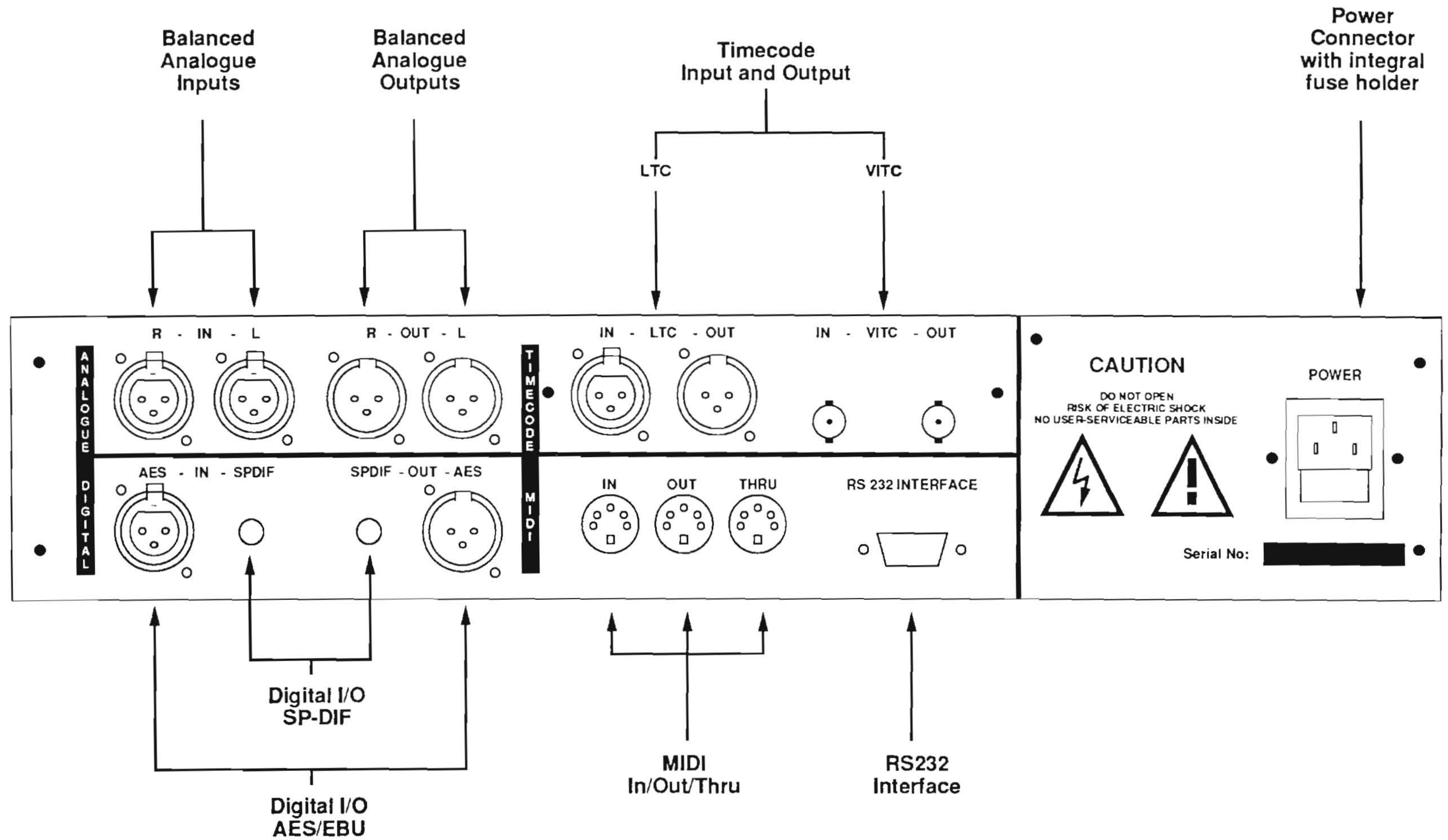
4. FREE STANDING USE

The DH-1 can be used as a free-standing unit. The rack-mount ears may then be replaced by the blanking plates if desired.

To replace the ears with the blanking plates:

- Unscrew the three bolts which attach each ear to the chassis of the DH-1.

- Attach the blanking plates using the same retaining bolts. Do not over-tighten these bolts as doing so may cause damage to the DH-1.



Balanced Analogue Inputs
Balanced Analogue Outputs
Timecode Input and Output
LTC
VITC
Power Connector with integral fuse holder
R - IN - L
R - OUT - L
ANALOGUE
TIMECODE
IN - LTC - OUT
IN - VITC - OUT
CAUTION
DO NOT OPEN
RISK OF ELECTRIC SHOCK
NO USER-SERVICEABLE PARTS INSIDE
POWER
DIGITAL
AES - IN - SPDIF
SPDIF - OUT - AES
MIDI
IN
OUT
THRU
RS 232 INTERFACE
Serial No: [redacted]
Digital I/O SP-DIF
Digital I/O AES/EBU
MIDI In/Out/Thru
RS232 Interface

CONNECTIONS

The DH-1 may be connected to most of the professional audio equipment currently available. Three types of audio input and output are provided (one analogue and two digital) and these will satisfy most users' interconnection requirements. Full descriptions of these connectors will be found later in the manual.

1. BEFORE CONNECTION

- To prevent problems and possible equipment damage, turn off the power to all equipment before making connections.
- Be sure to insert plugs firmly into sockets. Loose connections may cause hum and noise.
- When unplugging any lead, do so by grasping the plug, not the lead.

2. POWER CONNECTIONS

Ensure that the DH-1 is switched OFF before inserting the mains lead.

NOTE: Users with 2-pin mains supplies:
When the DH-1 is connected to other audio components, the AC hum of the unit may be increased or decreased by reversing the direction of the power connector in the socket. Check that the cord is in the favourable position ('in-phase') with respect to other audio devices in the chain. This will ensure that the best sound quality is obtained from your DH-1.

For further information on grounding and polarity consult a person familiar with studio grounding techniques.

3. SIGNAL LEAD CONNECTIONS

Refer to the Rear Panel diagram:

The DH-1 offers three audio connection standards: one analogue and two digital. These are:

- balanced analogue audio I/O
- digital SP-DIF format audio data
- digital AES/EBU format audio data

Note that the DH-1 always passes its output to all three signal outputs irrespective of the input used, but that the digital data will only be formatted for ***either*** *AES/EBU* **or** *SP-DIF, as defined by the user parameters.*

(i) Balanced analogue audio I/O (Pin 2 - 'hot')

This standard is used in professional audio equipment. Connect the output from your source to the balanced analogue inputs of the DH-1 using standard XLR plugs. You will require two such connections: one for each channel.

The balanced audio output may be used to connect the DH-1 directly to audio equipment such as mixing desks and professional recorders featuring balanced XLR inputs and outputs.

(ii) Digital SP-DIF format audio data

The SP-DIF format is used by domestic and semi-professional digital audio devices such as DAT machines, some ADCs, and some CD players. Both audio channels are carried along a single cable, so you may connect the SP-DIF output from your source to the SP-DIF input of the DH-1 using a single cable terminated with RCA (or 'phono') plugs.

The SP-DIF output of the DH-1 may be connected to the SP-DIF input of your recording device or external DAC.

(iii) Digital AES/EBU format audio data

The digital AES/EBU format is used by professional digital audio devices including mastering systems, DASH recorders, and high quality ADCs & DACs. Both channels of audio are carried along a single cable, so you may connect the AES/EBU output from your source to the AES/EBU input of the DH-1 using a single cable terminated with XLR plugs.

The AES/EBU output of the DH-1 may be connected to the AES/EBU input of your digital mixer, recording device or external DAC.

24-bit Digital data resolution:

The DH-1 features 24-bit input and output resolution whenever the AES/EBU digital input and output are utilised.

Dithering:

The DH-1 *SERIES 2* also features TPDF (Triangular Probability Density Function) dithering. This is applied to the digital data when the SP-DIF output format is selected. Dithering is always applied to the data presented to the DACs.

In order to fully comply with EMC regulations, this unit should be connected via its AES/EBU and/or analogue connectors. Metal-shelled XLR connectors should be used. We recommend using a good quality 'starquad' cable, with three cores connected to pins 1, 2 & 3. The shield of the cable should be connected, at both ends, to the outer shell of the connector.

4. OTHER CONNECTIONS

(i) **SMPTE/EBU**

An optional SMPTE/EBU interface offering LTC and VITC protocols is available for the DH-1. The standard DH-1 does not support timecode and these connectors are not present.

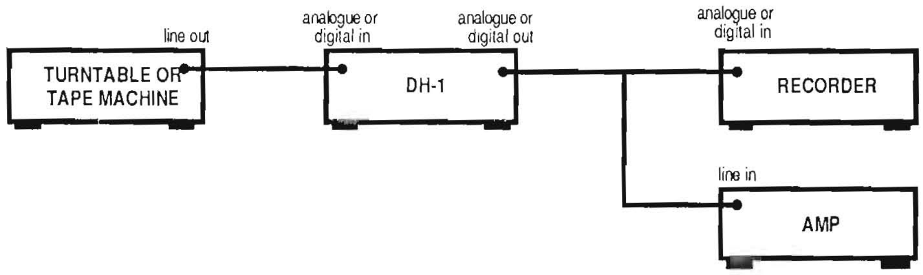
(ii) **MIDI IN/OUT/THRU**

The operation of the DH-1 may be controlled using the Musical Instrument Digital Interface (MIDI). Refer to the chapter on Remote Control Protocols for further instructions.

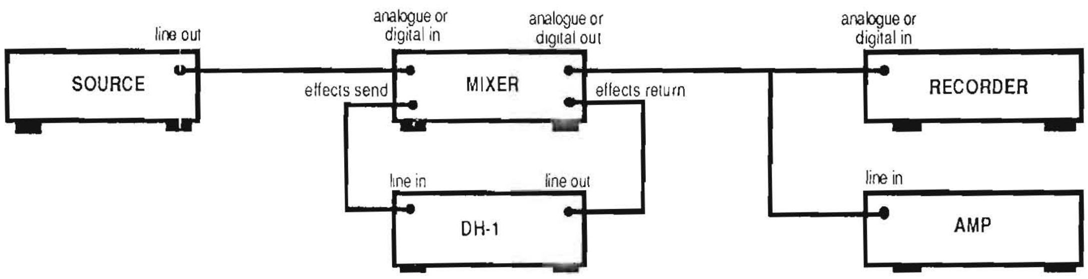
(iii) **RS232**

The DH-1 may be controlled using the standard RS232 serial communications protocol. Refer to the chapter on Remote Control Protocols for further instructions.

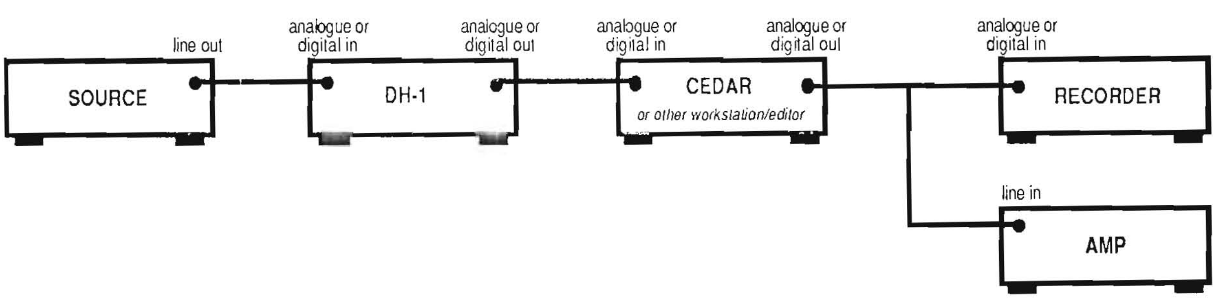
SAMPLE INSTALLATION IDEAS



1. *DH-1 used in-line for transcription or broadcast purposes.*



2. *DH-1 used on the effects loop within a studio environment.*



3. *DH-1 used in-line prior to an editor or audio workstation.*

A GUIDE TO RESTORATION PROCESSING

Contrary to 'common sense', the order in which restoration processes are carried out makes a great deal of difference to the quality of the final result. Consequently, there is one 'right way' and many 'wrong ways' to restore your material.

Following these guidelines will help you to achieve the best results on most material:

- De-Clicking (De-Scratching) should ALWAYS be carried out first. This is because:

  i Large clicks make it difficult for the De-Crackling process to identify and remove the tiny clicks and crackles that constitute surface noise, buzz, and other such problems.

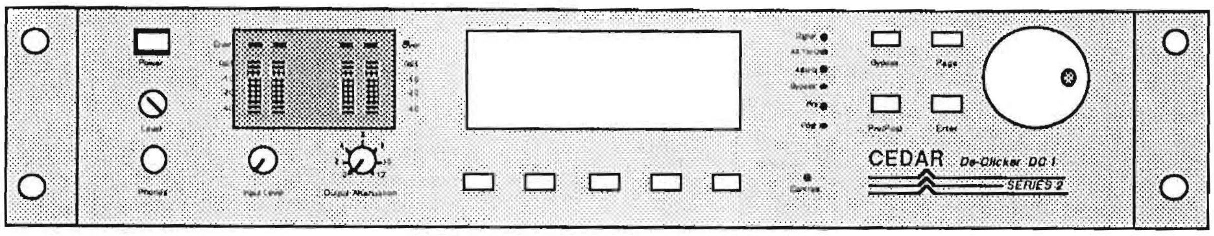
  ii All clicks and scratches are, in effect, tightly defined packets of white noise. If clicks are presented to any of the CEDAR De-Hiss products (HISS-1, HISS-2, DH-1 De-Hiss) they confuse the processes, and create unmusical side effects. In addition, De-Hissing at this stage will make it almost impossible to identify and remove clicks and scratches at a later time.

- De-Crackling should be the next process because even small crackles can cause the same problems as in (ii) above.

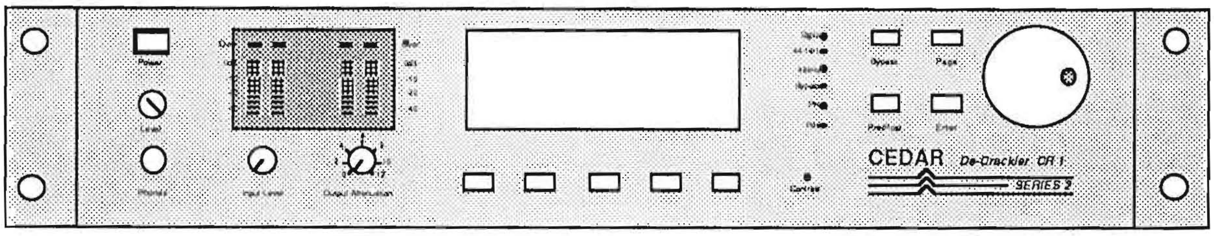
- Azimuth Correction can be carried out either before or after De-Hissing, but experience shows that best results are obtained using the AZ-1 or Phase-EX module before De-Hiss.

- Finally, apply whichever De-Hiss process you wish to use.

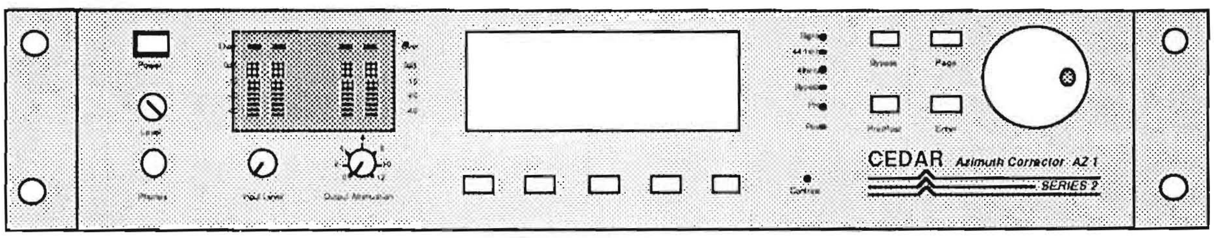
Note: If you have the full range of CEDAR restoration modules they should be connected as shown in the diagram overleaf. Please note that, to maintain the maximum fidelity and remove and possible sources of degradation between processes, connections between modules should be by AES/EBU (24-bit) format.



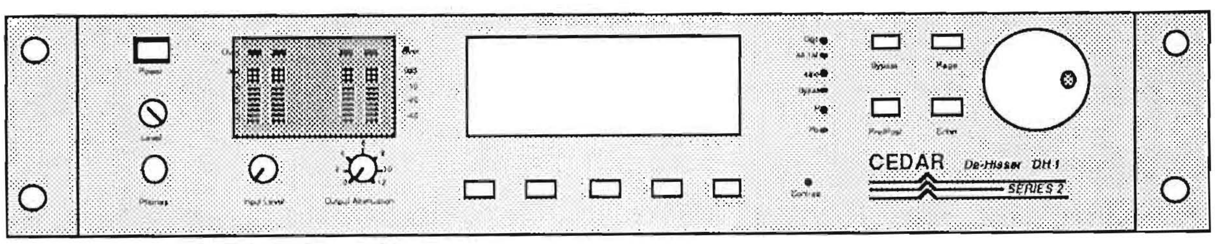
Firstly, De-Click your material



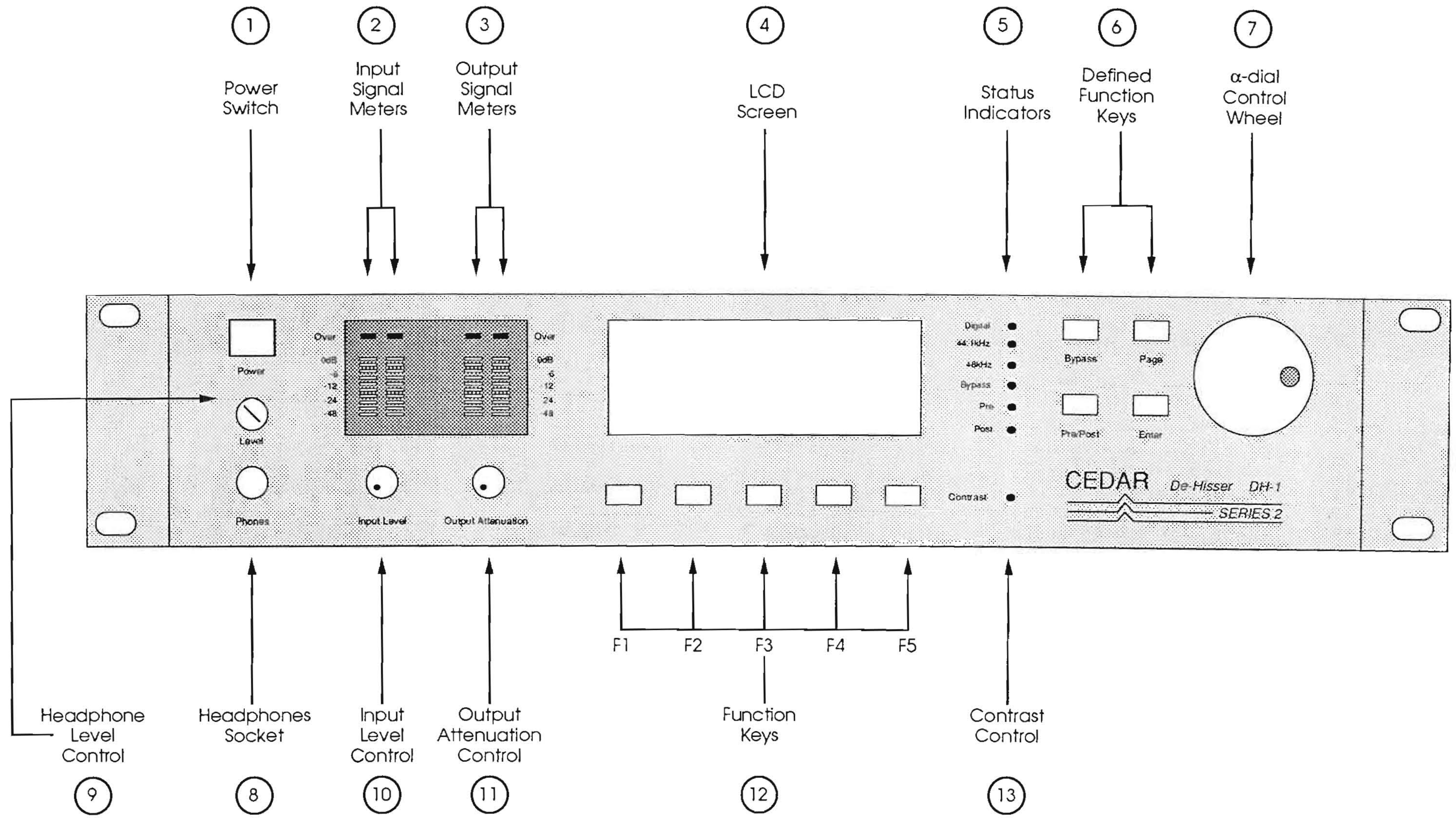
Next, remove crackle and buzz, and reduce distortion if appropriate



Then apply Azimuth Correction to material with phase and balance problems



Finally, apply noise reduction.



1
Power Switch
2
Input Signal Meters
3
Output Signal Meters
4
LCD Screen
5
Status Indicators
6
Defined Function Keys
7
α-dial Control Wheel
Power
Level
Phones
Input Level
Output Attenuation
Over
0dB
Digital
44.1kHz
48kHz
Bypass
Pre
Post
Contrast
Bypass
Page
Pre/Post
Enter
CEDAR De-Hisser DH-1
SERIES 2
F1
F2
F3
F4
F5
Headphone Level Control
Headphones Socket
Input Level Control
Output Attenuation Control
Function Keys
Contrast Control
9
8
10
11
12
13

LOCATION AND FUNCTION OF FRONT PANEL INDICATORS AND CONTROLS

Refer to the Front Panel diagram:

1. **Power Switch**

2. **Input Signal Meters (Left and Right)**
   Digital signal meters display the peak value of the selected input in dB0s.

   The 'Over' indicators will light if the input signal remains at full scale for four or more consecutive samples.

3. **Output Signal Meters (Left and Right)**
   Calibrated signal meters display the RMS value of all output signals.

   The 'Over' indicators will light if the output signal remains at full scale for four or more consecutive samples.

4. **LCD Screen**
   Provides you with a variety of information and messages, keeping you aware of what is currently happening in the DH-1.

   All the control screens of the DH-1 are displayed on the LCD screen. Please refer to the following chapters for full instructions.

5. **Status Indicators**
   Indicate the status of the analogue and digital inputs, and whether the DH-1 *SERIES 2* is in idle or processing modes.

   Also indicate the possible causes should the unit fail to function.

6. **Dedicated Function Keys.**
   Certain functions are fundamental to operating the DH-1, and these are controlled by the Dedicated function keys: Bypass, Page, Pre/Post, and Enter.

7. **α-dial (Spinwheel)**
   The α-dial enables you to increase and decrease control values. Please refer to the following chapters for full instructions.

8. **Headphone Socket**
   For use with stereo headphones only. Accepts a standard 1/4" stereo jack plug. DO NOT use 2-conductor mono headphones with the DH-1.

9. **Headphone Level Control**
   Use this to adjust for a satisfactory listening level. This level control will not alter the signal level at any of the rear panel outputs.

10. **Input Level Control**
This control acts upon the analogue inputs only. Use it to adjust the volume of incoming analogue signals to the desired level. A level of approximately 0 to -3dB (as shown on the Input Signal Meters) will offer best results.

The Input Level Control may be physically bypassed internally to obtain the best possible signal to noise ratio (S/N) from the ADCs. This work must be carried out by qualified service personnel, so please refer to your authorised dealer or directly to CEDAR Audio to have this modification performed.

11. **Output Attenuation Control**
A digital gain control with range 0 to -10dB in 1dB steps.

12. **Function Keys**
Use along with the LCD screen. Please refer to the following chapters for full instructions.

13. **Contrast Control**
The LCD screen may be adjusted for optimum visibility. Use a fine screwdriver to make such adjustments.

QUICK TOUR

If you are impatient to hear some immediate results using your DH-1 the following instructions should have you up and running within a few minutes:

1. **READ THE SAFETY INSTRUCTIONS.**

2. Connect the DH-1 to the mains supply.

3. Connect your input and output devices to the DH-1 using the appropriate input and output sockets. (If in doubt, please refer to the section CONNECTING THE DH-1 and the manuals of your other equipment).

4. Referring to the front panel diagram, hold down the function key F1 and switch on the DH-1.

5(i) If you are using analogue inputs press PAGE once. Press B (function key F2) to select 'analogue'. Then press PAGE twice more to return to the Control Page.

5(ii) If you are using digital inputs from a consumer format machine such as a domestic DAT recorder press PAGE once, then press B twice to select 'SP-DIF'. If you are outputting to a consumer format machine such as a low-cost DAT recorder press A (function key F1) to select SP-DIF format.

Press PAGE twice to return to the Control Page

Note: The DH-1 defaults to AES/EBU PROFESSIONAL format, so skip both instructions 5(i) and 5(ii) if your DH-1 is connected to a system such as the Sony PCM1630.

6. Play your material through the DH-1.

7. Press function key F2 to select the LEVEL control, and use the α-dial to vary LEVEL between 0.00 and 99.00. Provided that the material you are playing contains hiss you will, at some point within the scale, hear it disappear.

8. With ATTEN set to -40.0 and AMB set to 0.00 you will almost certainly hear side effects while you adjust LEVEL. Experiment with ATTEN and AMB to hear how these effect the output. Please refer to the TUTORIAL section for a full explanation of these controls, how they interact, and how to get the best results from them.

This section should have whetted your appetite, so you should now proceed to the rest of the manual.

WARMSTART AND COLDSTART

The DH-1 features Warmstart and Coldstart options. Warmstart has been added so that the unit can be configured once, and these parameters are then automatically recalled on every power-up. This is ideal for applications where time-consuming set-ups at the start of each session are not practical.

Coldstart

If the DH-1 has not been used for some time the system will automatically Coldstart. This process initialises all parameters to their factory default values, and after a few seconds the DH-1 will automatically enter at Page 1.

On start-up the message 'Coldstart' will be displayed at the top right of the start-up screen on the LCD display. The screen will then enter PAGE 1, which will show the default Parameters:

The default values are:	LEVEL	=	0.00
	ATTEN	=	-40.0
	AMB	=	0.00

Other default values are:	Digital Output	=	AES/EBU
	Input Source	=	AES/EBU
	Receiver Error Level	=	1 - Lock
	MIDI	=	Channel 1
	Bypass	=	OFF
	A to D frequency	=	44.1kHz
	Pre/Post	=	Post

Warmstart

The DH-1 remembers the latest parameters used, and the page that was active at the time that the system was last switched off.

On start-up the DH-1 will display the message 'Warmstart' on the screen, and after a few seconds will re-enter at the appropriate page, with all user parameters set to their previous values.

User Coldstart

If you wish to force the DH-1 to Coldstart, hold down Function Key F1 while switching on the system. Release F1 when the message Coldstart is seen on the LCD display.

*Note: In common with all other digital devices, and irrespective of whether you are Warmstarting or Coldstarting the DH-1, you should always allow a few seconds between switching the unit **off**, and switching it **on** again.*

OPERATING THE CEDAR DH-1

1. DEDICATED CONTROLS:

The DH-1 features a number of dedicated controls to speed operation. These are:

Dedicated Function Keys:

- Bypass
- Page
- Pre/Post
- Enter

I/O Level Controls

- Input Level
- Output Attenuation

These are now explained in turn:

Bypass

You may wish to bypass completely the operation of the DH-1. Press BYPASS to do this. The current status will be indicated on the Status LED.

The Bypass does not 'hard-wire' the input to the output. Analogue signals still pass through the A/D and D/A stages.

Notes:

- *There is a delay of approximately 1.3mS in any analogue-to-analogue signal passed through the DH-1 in Bypass mode.*
- *There is a delay of approximately 0.1mS in any digital-to-digital signal passed through the DH-1 in Bypass mode.*
- *All delays are 'group delays' (i.e. are constant at all frequencies) and are measured at a sample rate of 44.1kHz.*

Page

Use this Function Key to move between Pages.

Pre/Post

It will often be useful to compare the original signal with the post-processing output of the DH-1. The current status will be indicated on the Status LEDs.

Enter

The ENTER Key has three functions: as a LOCK-OUT key, preventing accidental changing of parameters; as a CLEAR key, resetting error messages, and as a MIDI DUMP command.

These first two functions are, of course, context sensitive, and the key's action will be appropriate to the page displayed (see below). The MIDI DUMP will be initiated every time that the ENTER key is pressed, regardless of context.

Input Level

This control acts upon the analogue inputs only. Use it to adjust the volume of incoming signals to the desired level. We recommend a peak level of approximately 0 to -3dB as shown on the Input Signal Meters.

Output Attenuation

Avoid clipping using the Output Attenuation Control. This is not a compressor or limiter, and acts purely as a digital gain control with variable gain from 0dB to -10dB in 1dB steps.

OPERATING THE CEDAR DH-1

2. PAGES:

The DH-1 has three 'pages' which control all aspects of its operation. Each page is displayed on the LCD screen, and may be controlled using the Function Keys and the α-dial.

Switch the DH-1 on. (Refer first to the safety instructions.)

The screen will immediately enter the CONTROL PAGE, which will show the Warmstart parameters stored when the unit was last used.

All the controls for the DH-1 are contained in the PAGES, each of which is selected by pressing the dedicated **PAGE** function key. The Pages are cycled, and will appear in the following order:

- Control Page
- I/O Control Page
- Remote Control Page

These, and a further description of the Dedicated Controls, are now covered in turn.

Note: *There is a fourth, normally hidden, page called the Status Page. This is not accessed using the standard 'Page' function, and will be discussed* separately in the section describing Error Levels.

PAGE 1: CONTROL PAGE

If necessary, access this page by pressing the Dedicated Function Key PAGE until the Control Page appears.

There are five controls in the Control page. These correspond to the five 'soft-keys' and are to be found directly above each of them as follows:

F1	•	Stereo Ganging/Left/Right Control
F2	•	Level Control
F3	•	Atten (Attenuation) Control
F4	•	Amb (Ambience) Control
F5	•	Clear

The controls, and therefore the DH-1 itself, act in the following manner:

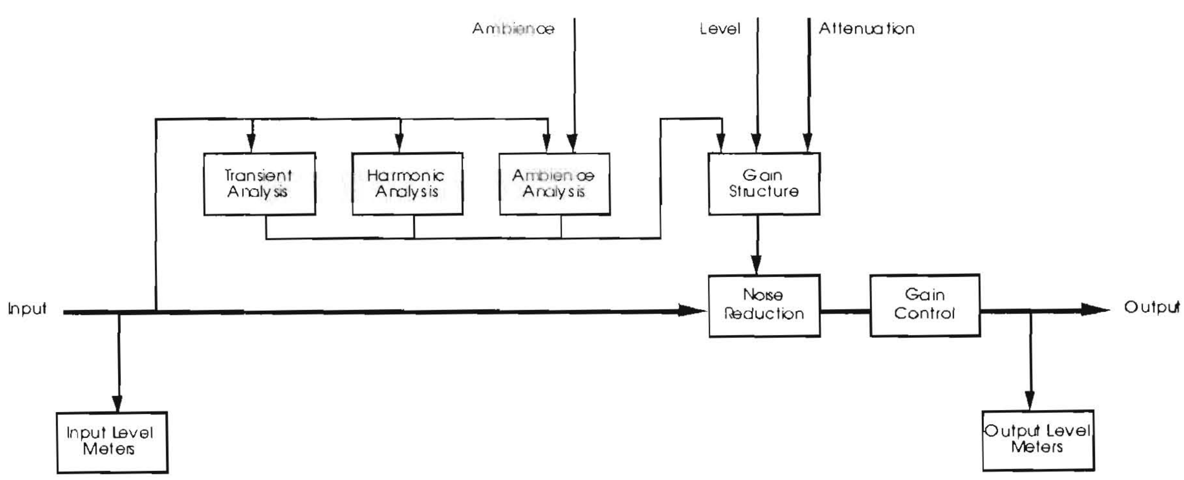


Figure 1: The DH-1 process overview, and the stages at which the CONTROL PAGE controls modify the signal.

You can optimise the beneficial effect of the DH-1 by setting each of these controls appropriately. They are now described in turn:

Level:

The LEVEL control is used to give the DH-1 algorithm a rough idea of the amount of noise present in any given signal. This is the most sensitive and important control on the DH-1, and incorrect use will result in sub-standard results and/or unwanted side-effects.

LEVEL may be adjusted as follows:

- Press F2 to select the LEVEL control. A box will appear around the numerical display to indicate that the control is selected.
- Rotate the α-dial clockwise and/or anti-clockwise to alter LEVEL in steps of 0.01.
- Rotating the α-dial slowly will result in delicate adjustments, whilst faster rotation will increase the rate at which LEVEL changes.

Atten:

ATTEN sets a maximum limit on the amount of noise that the DH-1 will remove at any time at any given frequency. It is quantified in dBs.

ATTEN should be used to ensure that no side-effects or loss of high frequencies caused by over- or under-processing are heard in the output signal.

It may be adjusted as follows:

- Press F3 to select the ATTEN control. A box will appear around the numerical display to indicate that the control is selected.
- Rotate the α-dial clockwise and/or anti-clockwise to alter ATTEN in steps of 0.1.
- Rotating the α-dial slowly will result in delicate adjustments, whilst faster rotation will increase the rate at which ATTEN changes.

Amb:

The 'speed' at which the noise attenuation is allowed to approach the ATTEN value is determined by the AMB (Ambience) control. This enables you to ensure that sounds are not prematurely cut short (hence the term ambience, implying the retention of natural decays and reverberations).

AMB may be adjusted as follows:

- Press F4 to select the AMB control. A box will appear around the numerical display to indicate that the control is selected.
- Rotate the α-dial clockwise and/or anti-clockwise to alter AMB in steps of 0.01.

- Rotating the α-dial slowly will result in delicate adjustments, whilst faster rotation will increase the rate at which AMB changes.

Ganging:

The DH-1 may be used to process stereo material. The left and right channels of such material can be entirely independent and exhibit quite different noise characteristics. The "Ganging" control allows you to select which channel(s) are affected when you adjust LEVEL, ATTEN, and AMB.

The Ganging control has three modes:

Ganged:	Adjusting the controls affects the left and right channels identically unless such adjustment would move a channel beyond the limits of the scale. In this mode, the numeric readouts beneath the control bars displays the average value of the left and right channels' values.
Left:	Only the left channel is affected, and the numeric readouts beneath the control bars displays the left channel's values.
Right:	Only the right channel is affected, and the numeric readouts beneath the control bars displays the right channel's values.

Press F1 to toggle between modes.

Clear:

The settings of LEVEL, ATTEN, AMB, and Ganging Controls may be returned to their default values simply by pressing CLEAR.

No other DH-1 controls or options are affected by this operation.

PAGE 2: INPUT/OUTPUT CONTROL PAGE (I/O CONTROL)

Access this page by repeatedly pressing the Dedicated Function Key PAGE until the I/O CONTROL PAGE appears.

This page allows you to determine the input used; the sampling frequency of the Analogue to Digital Converters; the digital input error detection level; and the digital output format.

(Remember that all outputs are permanently active, and that they do not require selecting, but that the same digital data is supplied to both AES/EBU and SP-DIF outputs. The data format will therefore only be appropriate for one digital output at any given time.)

There are three options in the I/O Control Page:

A. DIGITAL OUTPUT:

This option defaults to AES/EBU. To toggle between the two output modes, AES/EBU and SP-DIF, press the Function Key marked 'A' on the LCD screen.

- AES/EBU FORMAT:

  When AES/EBU is selected, both the phono and XLR connectors will carry AES/EBU specification audio data. You should patch the output from the XLR connectors to your recording device.

  The DH-1 features 24-bit input and output resolution when AES/EBU is selected.

- SP-DIF FORMAT:

  When SP-DIF is selected, both the phono and XLR connectors will carry SP-DIF specification audio data. You should patch the output from the phono connectors to your recording device.

  TPDF dithering will be applied to the digital data at the 16-bit level and is always applied at the analogue output.

B. INPUT SOURCE:

There are three input sources: AES/EBU, SP-DIF and ANALOGUE.

To toggle between the input sources press the Function Key marked 'B' on the LCD screen. The Status LEDs will indicate the inputs selected and the sample rate received (digital) or selected for conversion (analogue).

- SAMPLE RATE OF INCOMING DIGITAL SIGNAL:

  When the DH-1 is switched to receive digital audio data, the 'DIGITAL' LED will be lit, and the front panel LEDs will indicate the sample rate of the digital signal presented to the inputs:

neither 44.1 nor 48 kHz LED lit	=	32 kHz signal presented to inputs
44.1 kHz LED lit	=	44.1 kHz signal presented to inputs
48 kHz LED lit	=	48 kHz signal presented to inputs

- CLOCK DETECTION:

  If the DH-1 fails to detect a digital signal within the following limits, the 44.1kHz and 48kHz LEDs will flash continually. This will be irrespective of any other system settings.

Acceptable ranges:	44.1kHz	± 4%
	48kHz	± 4%
	32kHz	± 4%

- SAMPLE RATE OF A TO D CONVERTERS

  When the DH-1 is switched to receive analogue audio data, the 'DIGITAL' LED will not be lit, and the front panel LEDs will indicate the sample rate of the analogue-to-digital converters.

  The ADCs in the DH-1 do not offer a 32kHz option unless synchronised to an external 32kHz source.

C. A TO D FREQUENCY (INPUT SOURCE = ANALOGUE)

The ADC frequency may be selected by two, fundamentally different, methods. The first is to select one of the internal clock frequencies available, the second is to control the sample rate by using an external clock.

- INTERNAL CLOCK FREQUENCIES

  To toggle between the DH-1's internal 44.1kHz and 48kHz sampling frequencies (and between AES Sync and SP-DIF Sync - see below) press the Function Key marked 'C' on the LCD screen. The change in frequency will be shown on-screen and also by the Status LEDs.

  Note: The sampling frequency reverts to 44.1kHz on Coldstart.

- EXTERNAL SYNCHRONISATION

  The DH-1 clock may be synchronised to either the AES/EBU input or the SP-DIF input. Connecting a valid digital input to either of these and selecting AES Sync or SP-DIF Sync (as appropriate) will lock the DH-1 to the external clock.

  If the external clock falls within the acceptable ranges of each of the standard sample rates (44.1kHz, 48kHz, 32kHz) the clock frequency will be shown on the LEDs. If the external clock lies outside these ranges the DH-1 will still function, and good audio will be produced at the analogue output. Whether the digital output will be usable will then be determined by the flexibility of other devices in the digital audio chain.

  To toggle between AES Sync and SP-DIF Sync (and also between the internal 44.1kHz and 48kHz sampling frequencies) press the Function Key marked 'C' on the LCD screen.

  Note: If external synchronisation is requested, but no valid signal is detected at the appropriate digital input, the DIGITAL LED will flash to indicate the error.

D. RECEIVER ERROR LEVEL (INPUT SOURCE = AES/EBU or SP-DIF)

The DH-1 features sophisticated software which detects and analyses both fatal and non-fatal errors in the incoming digital audio data.

You may select one of four error levels which will cause the front panel 'DIGITAL' LED to flash if the incoming data contains an error equal to or worse than the selected level.

The error levels are:

- **1 - Lock**

  This is the 'weakest' detector and will only cause the LED to flash when the DH-1 believes that there is no usable signal being presented to the selected digital input.

- **2 - Code**

  If there is an incoming signal yet the LED flashes on error level 2, the DH-1 is indicating that the signal contains coding violations. In some cases you may obtain usable audio. However, this warning may be caused by non-AES/EBU or non-SP-DIF data being presented. In these cases any audio produced will almost certainly be unusable.

- **3 - Trans**

  This indicates that the incoming digital audio data is of poor quality (i.e very noisy or jittery) and that undetectable data errors are likely. These errors will not be corrected by any standard AES/EBU or SP-DIF device and may lead to audio degradations.

- **4 - Valid**

  This is the most stringent test of the incoming data, and will cause the LED to flash if the DH-1 determines that any of the data contained in the signal is not valid. This is often non-fatal (i.e. you will hear perfectly good audio) but it indicates that some device or anomaly in your audio chain is generating digital audio data outside of the AES/EBU or SP-DIF specifications published by their respective bodies. Please note however that, if the digital LED does not flash, this can not be taken as an absolute statement that the signal conforms to specification.

Note: If the error level selected detects an error, the digital audio signal will be coded as INVALID by the DH-1. Many manufacturers' devices do not recognise or act upon this code, but those that do may refuse to accept or record the audio.

PAGE 3: REMOTE CONTROL

Access this page by repeatedly pressing the Dedicated Function Key PAGE until the REMOTE CONTROL PAGE appears.

The DH-1 features intelligent 'auto-detection' software which monitors the RS232, MIDI, and SMPTE/EBU (if fitted) inputs and responds to data received on each and any of them. This eliminates the need for a control to select the remote control to be used.

It is only necessary, therefore, to select the Channel on which the DH-1 receives commands over MIDI.

MIDI

CEDAR Audio Ltd do not produce software for remote devices to control the DH-1 over MIDI.

- MIDI CHANNEL

  Ensure that button A is highlighted by a box. It is then possible to change the MIDI Channel turn the α-dial clockwise (to increase) or anti-clockwise (to decrease) the MIDI Channel.

  To toggle this function on/off press the Function Key marked 'A'.
  On Coldstart the MIDI Channel defaults to 1.

RS232

CEDAR Audio Ltd do not produce software for remote devices to control the DH-1 over RS232. However, for users wishing to implement their own control software, the RS232 Protocol is outlined in the chapter 'RS232 Protocol'.

SMPTE/EBU Timecode

A separate SMPTE/EBU reader/generator board may be purchased and fitted inside your DH-1. Please contact your dealer for details of this.

PAGE 4: STATUS PAGE

Access the STATUS PAGE by holding down Function Key F5 and then pressing the Dedicated Function Key PAGE.

Should the DH-1 fail to function, or appear to function incorrectly, there may be an error contained within the digital audio data received at the System's inputs. The Receiver Error Level (see above) will notify you when an error has occurred, but it will not tell you what it is. For many users, this information will be adequate, but the DH-1 is capable of reporting errors and other status information in more detail.

The STATUS PAGE will give you information regarding the current status of the DH-1, and will give you details regarding any errors which have occurred since the unit was switched on.

Three items of information will always be reported by the DH-1. These are:

• DSP1:	Status	Crashed / Timed Out / Running
• DSP2:	Status	Crashed / Timed Out / Running
• I/O:	Condition	Error / Emphasis, Sample Rate

If a remote control error is detected, a fourth field will appear:

• Comms:	Error	Illegal Checkbyte / Illegal Command Size

STATUS INDICATORS

The front panel LEDs will help to identify the possible cause if the unit fails to function. The following table lists all possible combinations of LED error indications:

LED flashing:		Condition:
Digital	-	the digital input violates the Receiver Error Level
	-	*or* no digital sync is present (if requested in I/O page)
44.1 *and* 48kHz	-	unknown sample rate received at inputs
Bypass/Pre/Post	-	One or both of the DSPs have crashed.

STATUS PAGE DEFINITIONS:

Crashed	The DH-1 DSPs are failing to function. The only recourse is to switch the unit off, wait for a few seconds, and then switch on again. If this error re-occurs please refer your DH-1 to an authorised service centre.
Timed Out	If, for any reason, the DH-1 drops out of real-time (fails to pass audio to the output) this error will be reported. This should only occur if a sample rate of greater than 50kHz is presented to one of the digital inputs. This error is non-fatal, and the DH-1 should continue to function normally after it has occurred.
Running	The DH-1 DSPs are functioning correctly and, moreover, have been doing so since the unit was switched on.
Error	If the DIGITAL LED is flashing the most serious error will be detailed at this point. Errors are fully detailed in the DH-1 Service Manual.
Emphasis	If no error is detected, the I/O status will display the Emphasis condition: • OFF The Emphasis bit is not set. The DAC de-emphasis will not be engaged. • 50/15 The Emphasis bit is set to 50/15 µS. The DAC de-emphasis will be engaged. • J17 (AES/EBU only) The Emphasis bit is set to CCITT J17. The DAC de-emphasis will not be engaged. • Unknown (AES/EBU only) The Emphasis status is not indicated. The DAC emphasis status will not be altered.
Sample Rate	If no digital data error is detected, the measured sample rate presented to the digital inputs will be displayed to the nearest 100Hz.
Illegal Checkbyte	The RS232 or MIDI has received a command packet containing an illegal checkbyte (byte2).
Illegal Command Type	The RS232 or MIDI has received a command packet containing an illegal command type (byte4).

TUTORIAL

One method for determining the correct values of the DH-1 noise removal controls is outlined below. Please note that the tutorial assumes that the material is stereo and exhibits virtually identical noise characteristics in each channel.

1. Ensure that the DH-1 is in POST and that BYPASS is OFF.

2. Select the Control Page and press CLEAR to reset the values of the DH-1's process controls to their defaults.

3. Your first task will be to find the most appropriate setting for the LEVEL control. This will be the single biggest influence on the quality of the processed signal.

   Starting with LEVEL at 0.00, use the α-dial to increase the value. First you will notice that very little happens. Then, at some point (which is defined by the nature of the noise contained within the signal) the level of hiss begins to decrease rapidly. At about the same level, you will probably notice the introduction of a side-effect known as 'twittering'. As you continue to increase LEVEL, the 'twittering' will begin to disappear, but you will cause the on-set of high frequency compression and another side-effect, sometimes called 'glugging'.

   Reduce the value of LEVEL back to the point at which the noise is reduced with minimum side-effects, and then further reduce the LEVEL by a small amount (which, ultimately, only experience can teach you to judge).

   Note: *The sensitivity of LEVEL is very high, and the setting should be controlled using the very accurate numerical readout, not the coarse 'fader' representation.*

   WARNING: ***If you are using the analogue inputs you must not adjust the front panel INPUT LEVEL CONTROL after you have found a suitable LEVEL. Such adjustment will, of course, alter the amount of noise being presented to the DH-1's processors, and make the initial LEVEL inaccurate.***

4. Some users prefer to adjust AMB next, whilst others adjust ATTEN. Ultimately, you will discover that the three main controls are closely related, and that, for best results, you will need to adjust each of them a number of times. However, in this tutorial we will adjust AMB next, followed by ATTEN.

5. With AMB at 0.00, the processed signal will lack ambience and, in all likelihood, suffer from side-effects. Correct adjustment of this control will restore the original ambience and reduce or eliminate the side-effects.

   Increase AMB from 0.00 until you just begin to hear noise 'pumping' at the end of loud notes. Do not worry that you have introduced another side-effect, because it will not be present in the correctly processed signal. It is merely a useful method for determining the correct level for the AMB and ATTEN controls.

Note: Adjustment of AMB cannot replace high frequencies lost through the action of LEVEL. These must be recovered by setting LEVEL and ATTEN correctly. However, AMB will affect 'twitters', transforming them under certain conditions into low-level noise artefacts.

6. You can now adjust the ATTEN control to determine the amount of noise removed.

   Increase ATTEN from -40.0 to 0.0, at which point you will hear that the processed signal is identical to the unprocessed. (This is because the ATTEN control is limiting the amount of noise removal to -0dB at every frequency - i.e. there is no effect.)

   You may now reduce ATTEN to a level defined by the material and your taste. However, you will notice that, if LEVEL is high, you can only reduce ATTEN by a few dBs before the on-set of side-effects such as loss of transients and loss of high frequencies. If LEVEL is low, ATTEN can be reduced further, but with reduced effect.

   As you reduce ATTEN you may notice one of two detrimental effects occurring:

   - If there were 'twitters' present after step (3), and if you reduce ATTEN beyond the optimal level for the specific material being processed, the twitters may be re-introduced as a form of high-frequency noise modulation.

   - If there was loss of high frequencies present after step (3), you will notice that this loss is gradually re-introduced as you decrease ATTEN.

7. It is unlikely that the values of the three controls are already optimised, so you should now return to step (3) and attempt to find a better value for LEVEL. Having done this you will, no doubt, wish to modify AMB and ATTEN further.

   Continued fine-tuning of these controls will lead to excellent noise removal with little or no side-effects. However, the DH-1 is not a magic wand, and it may not be possible to restore some (especially badly degraded) material beyond a certain point. Only experience will enable you to judge whether you have removed as much noise as possible without unacceptable consequences.

GENERAL OVERVIEW:

However you approach the de-hissing process, and in whatever order you choose to adjust the controls, you will probably find that the following rules apply for all material:

- If ATTEN is small, LEVEL can be set lower without introducing twitters or hiss modulation.

- If ATTEN is small, AMB can be set higher without introducing noise pumping.

Nevertheless, we would be grateful to receive feedback from our users regarding their experiences with the DH-1. Any suitable hints and tips will be included in this tutorial in later versions of the manual.

WARNING:

Some users may use commercial test CDs or signal generators to test the operation of components within their audio systems. The DH-1 will generate distortion if a digital FSD (full-scale deflection) sine wave is applied to either the AES/EBU or SP-DIF inputs. This does not imply that your DH-1 is faulty, and the effect should be ignored.

THE TUTORIAL TAPE

We have supplied you with a DAT containing three samples of hissy music. These samples may prove useful in helping you to learn the features and capabilities of your DH-1.

TRACK 1: MENDELSSOHN

This track is of generally high quality and does not have a particularly high level of hiss. It is, therefore, fairly simple to process and achieve excellent results. LEVEL does not need to be set very high, and this eliminates the danger of unwanted side-effects. ATTEN should be set more carefully because the track is quite dynamic and has many transients that could be damaged by over-processing. You will find that AMB will be very effective in ensuring that the 'air' is retained after processing.

Typical Values: LEVEL: 35 -> 38 ATTEN: -4 -> -8 AMB: 65 -> 68

TRACK 2: HORSLIPS

This track is of lower quality with a higher level of hiss. It is, therefore, somewhat more difficult to process than the Mendelssohn. It is quite tricky to find the correct setting for LEVEL because, as soon as the hiss begins to disappear, the first twitters are introduced. Consequently, you must (if possible) find a LEVEL between the hiss and the side effects. ATTEN and AMB can then be set as before and, provided that the LEVEL is not too high, there should be minimal loss of either ambience or brightness. Note that the introduction of the track will require heavier processing than the rest because the noise is much more noticeable before the louder passages.

The following settings are average values which will work reasonably well for the whole track:

Typical Values: LEVEL: 45 -> 52 ATTEN: -3.5 -> -6 AMB: 50 -> 65

TRACK 3: THE LUTON GIRLS' CHOIR

This track is of poor quality with a high level of hiss, and is the hardest of the three tracks to process. (In fact, it is often impossible to obtain a 'perfect' result on material this noisy.) Best results are obtained by processing *either* quite lightly, skimming just a little hiss from the original, *or* quite heavily, in order to disguise much of the surface noise contained in the original. Note that, although the track does not sound very bright, it is still necessary to set the AMB control carefully, because this will have a noticeable effect on the result.

Typical Values: LEVEL: 36 -> 48 ATTEN: -3 -> -4 AMB: 65 -> 70

REMOTE CONTROL PROTOCOLS

1. RS232

RS232 is defined in the DH-1 *SERIES 2* as:

9600 baud
8 bits data
1 stop bit
No parity

A command packet contains 6 bytes. These are:

byte 1:	channel number byte: must be 0xAF
byte 2:	Checkbyte. Fixed: must be 0x63
byte 3:	command number (see below)
byte 4:	Command type. Fixed: 0x07
byte 5:	command value HIGH byte
byte 6:	command value LOW byte

The HIGH and LOW bytes together form a signed integer.

Command Numbers:		**Command Values:**		
0xF7	Clear Errors command	Any value	=	Clear all error messages
0xF8	Select Page command	1	=	Control Page
		6	=	I/O Control Page
		7	=	Status Page
		15	=	Remote Control Page
		-1	=	Toggle between Pages
		Any other value	=	Refresh
0xF9	Pre/Post command	0	=	Pre
		1	=	Post
		-1	=	Toggle
		Any other value	=	Refresh
0xFA	Bypass command	0	=	Bypass OFF
		1	=	Bypass ON
		2	=	RESERVED VALUE
		3	=	RESERVED VALUE
		-1	=	Toggle
		Any other value	=	Refresh
0xC0	Digital Output Format	0x80	=	SP-DIF
		0x00	=	AES/EBU
		-1	=	Toggle
		Any other value	=	Refresh

0xC1	Input Source	0	=	Analogue
		1	=	SP-DIF
		2	=	AES/EBU
		-1	=	Toggle
		Any other value	=	Refresh
0xC2	A to D Frequency	0	=	44.1kHz
		1	=	48kHz
		2	=	SP-DIF Sync
		3	=	AES/EBU Sync
		-1	=	Toggle
		Any other value	=	Refresh
0xC3	Receiver Error Level	0	=	1 - Lock
		1	=	2 - Code
		2	=	3 - Trans
		3	=	4 - Valid
		-1	=	Toggle
		Any other value	=	Refresh
0x20	Set Left LEVEL	Any value	=	(Left LEVEL) x 100
0x30	Alter Left LEVEL	Any value	=	Δ (Left LEVEL) x 100
0x21	Set Right LEVEL	Any value	=	(Right LEVEL) x 100
0x31	Alter Right LEVEL	Any value	=	Δ (Right LEVEL) x 100
0x22	Set Left ATTEN	Any value	=	(Left ATTEN) x 100
0x32	Alter Left ATTEN	Any value	=	Δ (Left ATTEN) x 100
0x23	Set Right ATTEN	Any value	=	(Right ATTEN) x 100
0x33	Alter Right ATTEN	Any value	=	Δ (Right ATTEN) x 100
0x24	Set Left AMB	Any value	=	(Left AMB) x 100
0x34	Alter Left AMB	Any value	=	Δ (Left AMB) x 100
0x25	Set Right AMB	Any value	=	(Right AMB) x 100
0x35	Alter Right AMB	Any value	=	Δ (Right AMB) x 100

2. MIDI

The DH-1 is permanently set to transmit any change of control page parameters or Pre/Post state via MIDI except when such a change is initiated by an RS232 or MIDI command. Therefore, if a MIDI sequencer such as Cubase™, Notator™, or EditTrack™ is connected to the DH-1, it will receive a running history of the unit's operation.

If your sequencer and audio sources are able to send and receive timecode, then the DH-1's MIDI capability may be used as the basis for an automation system.

Note: The absolute parameter values are not transmitted or received, so the user must ensure that any changes are relative to a desired starting value which can be set using MIDI DUMP.

If a MIDI DUMP of all control page parameters and the Pre/Post state is required, pressing ENTER at any time will initiate the DUMP.

Additional MIDI Command

The DH-1 will receive LOCAL ON and LOCAL OFF commands.
The Status Page will notify you of the current state.
Both WARMSTART and COLDSTART always set LOCAL ON.

This command cannot be initiated from the front panel of the DH-1.

SELF TEST MODE

The DH-1 *SERIES 2* features a powerful self-test mode which enables the System to check the operation of each of its major sub-systems, plus all of the user controls.

To enter the self-test mode:
Switch on the DH-1 *SERIES 2* while holding down the ENTER key. The DH-1 will perform each test in turn, and you may move to the next test by pressing the ENTER key. Consequently, any test may be skipped by pressing the ENTER key.

Note: Whilst the SELF-TEST is in progress, the ENTER key will not initiate a MIDI DUMP.

ROUTINE 1: BUTTON TESTING ROUTINE

The DH-1 *SERIES 2* will invite you to press each of the Function Keys (except ENTER) and each of the Dedicated Function Keys. Pressing a key will cause the display to change from OFF to ON.

ROUTINE 2: ATTENUATION KNOB TEST

The DH-1 *SERIES 2* will invite you to turn the Attenuation knob to check that the value displayed on screen matches the position of the knob..

ROUTINE 3: α-dial (SPIN WHEEL) TEST

Rotate the α-dial to check that values change smoothly in both positive (clockwise) and negative (anti-clockwise) directions.

ROUTINE 4: LED TEST

The DH-1 *SERIES 2* will flash all six green LEDs.

ROUTINE 5: METER TEST

The DH-1 *SERIES 2* will invite you to turn the α-dial to vary the levels displayed by each of the four input and output meters in turn. Press ENTER to step to the next meter.

ROUTINE 6: DSP1 TEST

The DH-1 *SERIES 2* will test its DSPs and internal memory. Please wait for this test to complete.

- If the System is fully functional the screen will display the message: **"Memory passed"**.
- If a memory error is detected the screen will display the message: **"Memory error at: ....."**.
- If a DSP failure is detected the screen will display the message: **"DSP1 is not responding"**.

If you observe this message please repeat the self-test. If the message recurs please contact your dealer for assistance.

WARNING: *The DH-1 SERIES 2 contains no user-serviceable parts. DO NOT UNDER ANY CIRCUMSTANCES attempt to service your unit.*

ROUTINE 7: DSP2 TEST

As above.

TEST COMPETED

Your DH-1 *SERIES 2* will now prompt you to press ENTER one more time to return you to operating mode (whether all tests have been passed or not).

Some failures will not stop you from using the DH-1 *SERIES 2* successfully. However, consistent failures should be notified to your dealer or directly to CEDAR Audio Ltd.

CEDAR DH-1

Designed and Manufactured by

CEDAR Audio Ltd
5 Glisson Road
Cambridge
CB1 2HA

Tel: & Fax: +44 (0)223 464117

DECLARATION OF CONFORMITY CERTIFICATE

DATE OF ISSUE: 19 DECEMBER 1995

EQUIPMENT: CEDAR 'SERIES 2' DH-1 DE-HISSER

MANUFACTURER: CEDAR AUDIO LTD
ADDRESS: 9 CLIFTON COURT, CAMBRIDGE, CB1 4BN

THIS IS TO CERTIFY THAT THE AFOREMENTIONED EQUIPMENT FULLY CONFORMS TO THE PROTECTION REQUIREMENTS OF THE FOLLOWING EC COUNCIL DIRECTIVES: ON THE APPROXIMATION OF THE LAWS OF THE MEMBER STATES RELATING TO:

89/336/EEC ELECTROMAGNETIC COMPATIBILITY:

APPLICABLE STANDARDS: EN 50081-1:92
EN 50082-1:92

73/23/EEC LOW VOLTAGE EQUIPMENT:

APPLICABLE STANDARD: BSEN 60-065:1994

SIGNED: GORDON REID
POSITION: MANAGING DIRECTOR
DATE: 19 DECEMBER 1995