Prism Sound

Dream AD-1 Classic

Operation Manual

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1. INTRODUCTION

The ***Dream*** AD-1 ***Classic*** converter is intended for professional audio use in applications where a large dynamic range is required; for example in the production of high quality 16-bit CD masters - where noise shaping can provide a weighted dynamic range of more than 110dB, or for conditions such as live recording where a large amount of headroom may be required.

In addition to the 20-bit output mode the unit can provide a 16-Bit output with optional noise shaping. This has a weighted dynamic range just short of the 20-bit output mode, but unweighted noise is significantly increased.

The Prism *Dynamic Range Enhancement (DRE)* system is also incorporated for use with 16-bit media where the requirement for decoding is not a disadvantage, such as for the temporary storage of high dynamic range material before transfer to a 20-bit processing stage. Use of DRE allows an increased dynamic range to be carried in a 16-bit data channel without the high levels of high frequency noise that noise shaping produces. This function provides virtually the full dynamic range of the converter in 16 data bits. The ***Dream*** AD-1 ***Classic*** provides a decode function to allow an encoded DRE signal to be converted back to either a 20-bit or a noise shaped or dithered 16-bit linear output.

This document describes the product and a series of application diagrams (PRAN065/1 sheets 1-6) are included at the back which show some of the many ways in which the AD-1 can be used. A set of FFT (spectrum analysis) plots are also included at the back of this manual showing the performance of the various operating modes and the Dynamic Range Enhancement systems incorporated in the product.

The product is available in several variants; this variant is the `AD-1 Classic'. For further information contact Prism Sound, +44 (0)223 424988 (fax +44 (0)223 425023)

1.1. Revision History

October 13, 1992	Version	P1.0
October 14, 1992		P1.1
June 14, 1993		P1.2
July 10, 1993		V1.3
September 14, 1993		V1.4

2. GETTING STARTED

This section deals with three subjects; the first is un-packing and checking that you have all of the items that are on the inventory below. The second deals with connecting up and starting work and the third explains in more detail the capabilities and applications of the AD-1.

2.1. Un-packing your AD-1

Check that you have the following items and that they are undamaged:

- AD-1 Classic A-D converter unit
- A copy of this manual
- 2-off BNC to RCA co-axial adaptors
- Mains lead

Check that you AD-1 C carries a label on the rear panel indicating the correct mains voltage for your application area and that the mains lead is of the correct type. If not, DO NOT CONNECT THE MAINS SUPPLY, but contact your distributor.

Keep the packaging for re-use in the event that the unit should be shipped to another location or in the event that it should ever need to be returned to the manufacturer for repair.

2.2. Using the AD-1 for the first time

To set up your AD-1 quickly for A-D conversion, set the input gain switches (viewed from the rear) to:

1	Up
2	Down
3	Up
4	Up
5	Up
6	Down
7	Up
8	Up

Set the Digital Signal Processing control switches on the rear to

1	Up
2	Up
3	Up
4	Up
5	Up
6	Down
7	Down
8	Up

If you have only a 16-bit recorder or monitoring system set SW2 `Down'.

Connect an analogue source to the analogue inputs (the above switch settings set digital full-scale modulation to +18dBu); these are wired as:

XLR - Female Left & Right Channel Inputs:

pin 1 Chassis & Mains earth
pin 2 Balanced input (Hot or `+')
pin 3 Balanced input (Cold or `-') or Screen if unbalanced source

Connect the digital output to a suitable destination such as a DAT or Hard-Disk recorder and arrange to monitor the source of the destination device. (Typically set your recorder into `Record & Pause' mode and monitor at its D-A converters.)

Connect the mains supply and switch on the AD-1 unit.

The front-panel LED's should all illuminate momentarily; if any of the LED's fails to light it may be faulty. If so, repeat the check and if the problem is still evident contact your distributor.

LED's will then revert to the operational state which for the above conditions would be:

Power	ON
Local ref	ON (providing no external ref. connected)
44.1	ON
48.0	ON
20-bit	ON
Noise Shaping	OFF (dither is set flat in the above list)
A-D	ON

Meters	Dependent on signal level

Front-panel controls LED's (if fitted)

D-D	OFF	
16	OFF	(unless DSP SW2 set down as above, or this front-panel switch is operated)
DRE	OFF	
DEC	OFF	

Check that variation of the input signal level produces some indication on the level-meter LED's; at the settings suggested above it will be necessary to provide +18dBu** to clip the converter and illuminate the red overload indicators. The signal present indicator will light for signals exceeding -30dBu** and the mid-level indicator will illuminate for signals exceeding +6dBu**.

** Note all levels mentioned here are subject to the +18dBu : 0dBFS gain setting see section 5.1 for more details.

Try changing the output word-length; press the <16> key on the front of the AD-1 to toggle between 16 and 20-bit output. To detect the effect you would need a very low input level signal (about -70 to -80 dBFS) and a substantial amount of gain applied in the digital domain after the AD-1 output; great care should be exercised when doing this as damage (or injury) may occur when very high gains are used. Alternatively, use an FFT analyzer such as the Prism Sound Dscope.

The switch settings above select flat dither as the linearization method for 16-bit output. SNS noise-shaping can be used and if preferred this should be left set on the DSP switches. DSP switch SW5 selects between dither and SNS.

2.3. AD-1 product concept and capabilities

The Dream AD-1 is a more complex and sophisticated product than other A-D converters because it provides several functions in one unit.

Primarily, it provides a very high quality A-D converter, but two other important features are also provided. These are Super-Noise-Shaping (SNS) and Dynamic Range Enhancement (DRE).

SNS is a process designed to reduce the perceived noise-floor of CD to virtually inaudible levels when the reproduction level is roughly the same at the listing position as at the principal microphone location for the original recording. It involves complex signal processing that modifies the spectrum of the noise-floor in accord with the inverse of the Fletcher - Monsen curve. The process is completely compatible with existing CD players; however best performance will be obtained (the benefit of SNS) with high-performance D/A converters with 18-bit or better dynamic range.

The Fletcher - Monsen curve shows the sensitivity of the ear over the audio frequency spectrum. Super-Noise-Shaping results in a noise penalty (see the FFT plots at the back of this manual) in the high-frequency end of the audio band above 15KHz and is suitable only at the final mastering stage prior to transfer to the 16-bit master that will be used by the pressing plant. For this reason, as well as the A-D mode the SNS processing can also be applied in D-D mode with 20-bit digital input and 16-bit digital output.

DRE is a process designed for quite a different purpose. This process is **NOT** compatible with existing players. It is designed for use with DAT or other existing 16-bit recorders (such as CD-R or 1630+U-matic) but requires a decode process on playback.

DRE is intended for applications where 20-bit dynamic performance is desired of the recorder but only a 16-bit recorder is available. DRE does not add noise like SNS but it does require decoding before sending to D/A converters for playback.

The plots at the back of this manual illustrate the various operating modes and the difference between DRE and SNS is illustrated.

The AD-1 provides both DRE coder and decoder; the coder can be used either with the A-D converter input **OR** with the digital input (D-D mode), so that 20 bit digital material can be reduced for 16-bit recording. The decoder can be used only with the digital input. It can have 20-bit linear output (probably the most common mode of use) or the <16> selection can be used in conjunction with the dither/SNS selector (DSP SW5) to set 16-bit dithered or 16-bit noise-shaped output.

At the back of this manual are a series of diagrams showing various applications and operating modes. Study these in conjunction with sections 6.13 and 6.14.

3. SPECIFICATION

3.1. Analogue Performance

These performance figures are as measured on the prototype units using 20-bit mode at 48kHz. DRE and noise-shaping were disabled, and the input level for full scale output was set to +18dBu unless otherwise mentioned.

Tests are performed to AES17-1991 (ANSI S4.51-1991) except where otherwise stated.

Input Impedance	20 kS
Alias rejection	>80dB above 28kHz (AES 17 requires a graph)
Input for full scale amplitude	Fixed levels +28dBu, +18dBu, +12dBu (±0.2dB) Variable +8dBu to +24dBu
Maximum input amplitude	+28dBu
Input gain stability	<0.01dB 1 hour
Frequency response deviation	+0.1 -0.3dB from 1 Hz to 20 kHz
Maximum level vs frequency	0dBFS from 10Hz to 20kHz with <-80dB THD+N
Inter-channel phase deviation	< 1E from 10Hz to 20kHz
Polarity	Non-inverting
Level-dependent gain linearity	< 0.001dB
Idle channel noise	-115dB unweighted (AES17 requires weighted)
Noise with signal	-115dB unweighted (THD+N at -60dBFS, 20-20 kHz, note: AES17 requires weighted noise figure)

THD+N at 997 Hz, -1dBFS -94dB

Note: At typical mid-program levels say -20dB to -40dB THD+N for a 20-bit converter is significantly better than for a 16-bit unit

Typical 16-bit at -30dB	: -63dB
Dream AD-1C at -30dB (20-bit mode)	: -83dB

Linear cross-talk <-90dB from 20 Hz to 20 kHz
note: AES17 requires a graph

Common mode rejection >70dB (20-1 kHz)
>50dB (20-10 kHz)

3.2. Delay

3.2.1 ADC Mode 54 samples (48kHz = 1.125ms, 44.1kHz = 1.225ms)

3.2.2 D-D mode 5 samples (48kHz = 104µs, 44.1kHz = 113µs)

3.3. Synchronization

3.3.1 Internal

Sampling Frequency: 44.1 or 48 kHz (+/- 10ppm)

3.3.2 External

Sampling Frequency: 30 to 50 kHz on AES3, optical or coaxial AES3 style input.
44.1 or 48 kHz ±1000ppm on SDIF-2 word clock input.

Digital Output Phase: As AES11 spec. (Matched to input phase within 5%

of a sampling period.)

Synchronization : 1. Digital Audio Reference to AES11 format on XLR (110ohm balanced) or BNC (75ohm unbalanced) digital input. Either of these inputs may be used as the digital input in D-D mode)

2. Unbalanced 75S SDIF-2 word clock to BNC input; on the BNC input either TTL-compatible square-wave Word-clock or AES/EBU synchronizing signals may be connected. The AD-1 automatically detects the signal type.

3.4. Digital Output

3.4.1 Format:

XLR	Professional, AES3 (110 S balanced)
BNC	Consumer, SPDIF/IEC958/CP340 (75 S coaxial)
Optical	Consumer, CP340 (TOSLink)

(SDIF-2 coaxial output option is available)

3.4.2 Word length: Either 16 or 20 bits, dithered to eliminate truncation distortion.

3.4.3 Channel Status: Appropriate Channel Status for professional format (AES3) is set on the XLR connector and consumer format (IEC958) is set on the on the coaxial output.

An internal link (see section 7.1) can select professional status for both outputs simultaneously; useful when 75-ohm co-axial format is used for transmission of AES/EBU signals.

SCMS is **NOT** asserted on the consumer output.

Custom channel status implementations are available to order; for example text labels or modified category codes can be provided.

[Prism Sound can also assist in the diagnosis and solution of Digital Audio

Interface problems using the DAS-90 digital audio analyzer and the DSA hand-held AES signal analyzer.]

The following is taken from the Prism Sound DAS-90/Dscope system showing the Channel Status output of the AD-1 when internally synchronized at 48.0kHz:

```
***************************************************
* Prism Sound DSCOPE v2.01 Digital Audio Analyzer *
***************************************************
 Received Channel Status settings - professional:
 ---------------------------------
Byte 0;       Bit 0:            1:PROFESSIONAL                  Bit 1:  0:AUDIO USE

              Bits 4-2:     000:EMP UNDEFINED           Bit 5:        0:SRC Fs LOCK
              Bits 7-6:     10:Fs = 48.0kHz
Byte 1;              Bits 3-0:     0000:CH-MODE UNDEFIND            Bits 7-4:  0000:NO
USER INFO INDIC.
Byte 2;              Bits 2-0:     000:AUX U/D 20-bit         Bits  7-3:     00000:(not
indicated)
Byte 3;              Bits 7-0:     00000000:(byte 1 vector target)
Byte 4;              Bits 7-0:     00000000:NOT REF
Byte 5;              Bits 7-0:     00000000:Reserved
Bytes 6-9;                  SOURCE ID=    "AD-1"
Bytes 10-13;                DESTINATION ID=       "    "
Bytes 14-17;                LOC T-C:00:00:00:00
Bytes 18-21;                DAY T-C:00:00:00:00
Byte 22;      Bits 3-0:     0000:(rsvd)                 Bits 7-4:  1100:*UNREL
Byte 23;                    CRC=CORRECT

 Received Channel Status settings - consumer:
 ---------------------------------
Byte  0;      Bit 0:            0:CONSUMER          Bit 1:                0:AUDIO USE
              Bit 2:            1:COPY PERMIT              Bits 4-3:      0 0 : N O
EMPHASIS
              Bit 5:            0:2-CHANNEL         Bits 7-6:     00:Mode 0
Byte  1;      Bits 7-0:     10010110:CATEGORY CODE:     A/D CONVERTER ( (C) INFO )
                            SCMS: Copyright not asserted; Original material

Byte  2;      Bits 3-0:     0000:NO SOURCE NUMBER       Bits 7-4:     0010:CHANNEL NUMBER
B
Byte  3;      Bits 3-0:     0010:Fs=48.0K               Bits 5-4:     01:ACCURACY I
              Bits 7-6:     00:(rsvd)
Bytes 4-23, all reserved, bits 7-0:
  [ 4]  00000000  [ 5]  00000000  [ 6]  00000000  [ 7]  00000000
  [ 8]  00000000  [ 9]  00000000  [10]  00000000  [11]  00000000
  [12]  00000000  [13]  00000000  [14]  00000000  [15]  00000000
  [16]  00000000  [17]  00000000  [18]  00000000  [19]  00000000
  [20]  00000000  [21]  00000000  [22]  00000000  [23]  00000000
```

Various data in the Channel Status will change according to prevailing conditions during normal use. In particular the output word length, sampling frequency, and emphasis flags may change. The unit can indicate `unlocked' on the professional output only if external lock (mode switch 7 down) is selected and the unit is unlocked.

3.5. Power

3.5.1 Mains voltage: Internally set for 90-120V (Brown and Red transformer connections) or 195-250V operation. (Brown and Orange transformer connections). The selected supply voltage is indicated on the rear panel.

3.5.2 Consumption: <25W

3.6. Physical Dimensions

19 inch (483mm) width (rack-mountable)
1U (44mm) high
10.25 inches (260mm) deep (including protruding BNC connectors to rear)

4. CONNECTIONS

Viewed from the rear the connectors are as follows:

Left	XLR-F	Analogue In (Left/A)
	XLR-F	Analogue In (Right/B)
	XLR-F	Digital In (AES3)
	BNC-75S	Digital In (CP340/IEC958) or Word Clock In (SDIF-2)
	OPTICAL INPUT	Digital In (CP340)
	BNC-75S	SDIF-2 output Left channel (CH1) *
	BNC-75S	SDIF-2 output Right channel (CH2) *
	BNC-75S	SDIF-2 Wordclock output *
	OPTICAL OUTPUT	Digital Out (CP340 - consumer)
	BNC-75S	Digital Out (IEC958 - consumer)
Right	XLR-M	Digital Out (AES3)

* Note: SDIF-2 outputs are an option which must be ordered separately; however SDIF-2 connectors are fitted as standard. The SDIF-2 option includes the internal expansion PCB which provides the necessary electronics to drive this format.

XLR wiring conventions, for all signals analogue and digital are:

- pin 1 Chassis & Mains earth
- pin 2 Balanced input or output (Hot or `+')
- pin 3 Balanced input or output (Cold or `-')

5. OPERATIONAL CONTROLS

There are two 8 lever DIP switch banks accessible from the rear of the unit. These controls are described as viewed from the rear of the unit. The left hand switch bank also has two screw driver controls, one each side.

5.1. Input level controls

The analogue level corresponding to maximum digital level (0dBFS) can be adjusted to either standard settings or with a continuously variable control. The switch settings are shown in the table below:

Level	1	2	3	4	5	6	7	8
+12dBu	Up	Up	Down	Up	Up	Up	Down	Up
+18dBu	Up	Down	Up	Up	Up	Down	Up	Up
+28dBu	Down	Up	Up	Up	Down	Up	Up	Up
Variable	Up	Up	Up	Down	Up	Up	Up	Down

The switches are in two banks, switches 1 to 4 for the left channel and 5 to 8 for the right channel. Only one switch per bank should be down. The variable setting allows adjustment with the screw-driver potentiometers to either side of the switch bank (the control next to switch 1 is for the left channel).

The analogue 0V can be linked to cable screen via the XLR pin 1; see section 7.2.

5.2. DSP System settings and default control settings

The right hand bank of switches controls the signal processing and sampling frequency synchronization of the unit. Switches 1-4 set functions which are also be controlled from the front-panel switches, if fitted. The initial (power-on) states for these functions are set by the rear-panel switches.

The rear-mounted switches continue to operate after initialisation but can be overridden by any subsequent operation of the front-panel controls.

The switch functions are shown in the table:

Switch	Up	Down
1	ADC	Digital input (see SW8)
2	20-bit output	16-bit output
3	DRE-encode	DRE-decode
4	Linear PCM	DRE function as SW3
5	Flat dither	Noise-shaping
6	IntFS = 48 kHz	IntFS = 44.1 kHz
7	Internal lock (at IntFS)	External lock (if present)
8	XLR lock reference (or DI)	BNC or Optical lock reference (or DI)

5.2.1 Switch 1 determines the signal source. If `Up' the analogue input is selected (ADC mode); if `Down' the digital source (according to the state of switch 8, see below) is selected. When switching between these modes the output signal is muted for a few seconds while the system re-configures.

5.2.2 The output word length is selected by switch 2. (Note: This selection must match the input word length of the device connected to the AD-1 output otherwise truncation distortion or an increased noise level will result.)

5.2.3 Switch 3 is used to set the mode of the DRE system; encoding is selected when up and decoding is selected when down. The DRE system must be activated by SW4 (down) otherwise SW3 is ignored.

When DRE encoding is used in conjunction with a 16-bit recorder (or transmission system), it yields virtually 20-bit performance. The noise floor in this mode has a gradually rising characteristic above about 8KHz and rises to approximately the same level as the 16-bit noise floor at about 22KHz. This is significantly better than Super-Noise-Shaping in the DC to 8kHz band and further, the HF noise penalty of noise-shaping is avoided. This makes the DRE system more appropriate when further editing, EQ-ing or other post-production tasks are yet to be performed. Plots of the performance of the encode-decode chain are provided at the back of this manual.

5.2.4 Switch 4 selects between linear (normal) PCM data format and DRE encode/decode processing.

5.2.5 Super-Noise-Shaping (SNS) or flat dither is selected by SW5. This produces up to 17dB of improvement in weighted noise (for 16-bit output, see section 5.2.2) with an unweighted noise penalty (mainly in the band from 15KHz to Fs/2) of up to 19dB. SNS can also be used in the 20-bit output mode in which case a small improvement to the 20-bit (weighted) noise-floor can be achieved.

Note: In D-D mode the dither and SNS processing is automatically muted if the incoming data does not require it. This keeps data unchanged if it is of the correct word length already. (For example a 20 bit to 20 bit or a 16 bit to 16 bit transfer.) This also results in a `digital black' (all zeroes) if fed with digital black source. The automatic dither muting will activate if the bits at the input that extend beyond the output word (ie those that would be truncated) are inactive for more than about 4000 samples (0.1 seconds). To protect against truncation distortion it will turn it back on the instant that the active word length increases.

5.2.6 Switch 6 selects the internal sampling frequency. This has no effect when locked to an external source.

5.2.7 Switch 7 is used to select an internal or external synchronisation source.

5.2.8 Switch 8 is used to set which digital input connector is in use. An AES11 (or AES3) synchronizing signal can be applied to the XLR Digital Audio input connector. Alternatively either and AES/EBU or SPDIF signal or a TTL-compatible Wordclock signal can be applied to the BNC DI connector.

The optical input is selected if switch 8 is down and there is no connection at the BNC input. The coaxial input is selected if switch 8 is down and the is no connection at the optical input. If both are connected and switch 8 is down then either may be selected.

NOTE : SDIF-2 WORDCLOCK SYNC source cannot be used in D-D mode.

6. FRONT-PANEL INDICATORS AND CONTROLS

Several front-panel controls are provided which can be used instead of the rear-mounted switches if fixed function is not the usual mode of service.

The front-panel buttons are optional and if fitted simply override the rear-panel controls which then serve only to set the initial state of the ***Dream*** AD-1 ***Classic*** at power on.

The front-panel buttons are explained below and at the end of this section a comprehensive table explains operating modes in the context of applications. Frequent reference to the set of diagrams (PRAN065/1 sheets 1 to 6) will aid understanding of the following sections.

6.1. Power indicator

This illuminates when the unit is powered.

6.2. Local Ref indicator

This illuminates when the unit is locked to the internal source. This will occur if internal lock is selected manually or if the unit cannot lock to the selected external source.

6.3. 44.1 and 48 indicators

These indicate the approximate operating sampling frequency. (Both are illuminated for external 32kHz operation)

6.4. 20-bit indicator

Indicates 20-bit linear PCM output is set. DRE encoding to DAT will NOT illuminate the indicator.

6.5. Noise-Shaping indicator

Indicates that Super-Noise-Shaping is selected on the output, if output is linear PCM. DRE decoding with 16-bit output and SNS will illuminate the LED but DRE encoding will not, irrespective of SW5.

6.6. A-D indicator

Indicates that the signal source is analogue if lit, digital if not.

6.7. Left channel metering

Red:	Signal clip	(Signal > -0.03dBFS) This holds for approximately 2 seconds.
Orange:	Signal line-up level	(Signal > -12.04dBFS)
Green:	Signal present	(Signal > -48.16dBFS)

6.8. Right channel metering

Level indication as for left channel metering

6.9. Front-panel D-D button

The electronically latching D-D button selects the digital input mode when the LED is lit, analogue input otherwise.

6.10. Front-panel 16-bit button

The electronically latching 16-bit button selects 16-bit output word-length when a linear PCM output mode is active (when a DRE function is not selected).

6.11. Front-panel DRE button

The electronically latching DRE button selects the DRE encode function. In this mode, word-length and dither selections are not adjustable and indicator LED's will be off.

6.12. Front-panel DEC button

The electronically latching DEC button selects the DRE decode function. Word-length and dither selections are important and will depend on the word-length of the device to which the AD-1 output is connected.

6.13. A-D operating modes

In the following table the symbol `"' indicates an LED unlit and `•' an LED lit. In the table of rear panel switch settings 1 implies the switch is in the `up' position and 0 the `down' position.

Front panel control settings:	Rear Pnl Sw 1234 5678 *	Application
MODE 1 Linear, 16-bit, flat dither Buttons: [D-D "], [16 •], [DRE "], [DEC "] Leds : 20("), N-S ("), A-D(•)	[10x1] 1xxx	Record to 16-bit media with TPDF (flat) dither, achieving 93dB SNR.
MODE 2 Linear, 20-bit, flat dither Buttons: [D-D "], [16 "], [DRE "], [DEC "] Leds : 20(•), N-S ("), A-D(•)	[11x1] 1xxx	Record to 20-bit media with TPDF (flat) dither, achieving 115dB SNR.
MODE 3 Linear, 16-bit, Super-Noise-Shaped Buttons: [D-D "], [16 •], [DRE "], [DEC "] Leds : 20("), N-S (•), A-D(•)	[10x1] 0xxx	Record to 16-bit media with super noise-shaping.
MODE 4 Linear, 20-bit, noise-shaped Buttons: [D-D "],[16"], [DRE "], [DEC "] Leds : 20(•), N-S (•), A-D(•)	[11x1] 0xxx	Record to 20-bit media with noise-shaping.
MODE 5 DRE. Buttons: [D-D "], [16 "], [DRE •], [DEC "] Leds : 20("), N-S ("), A-D(•)	[1x10] xxxx	DRE Encode to 16-bit media.

<u>*NOTE:</u> Rear panel SW1-4 set power-on defaults ONLY

6.14. D-D operating modes

In the following table the symbol `"' indicates an LED unlit and `•' an LED lit. In the table of rear panel switch settings 1 implies the switch is in the `up' position and 0 the `down' position.

Front panel control settings:	Rear Pnl Sw * 1234 5678	Application
MODE 6 Linear, 16-bit, Super-Noise-Shaped Buttons: [D-D •], [16 •], [DRE "], [DEC "] Leds : 20("), N-S (•), A-D(")	[00x1] 0xxx	D-D transfer to 16-bit media with super noise-shaping for CD.
MODE 7 UnDRE, 16-bit, Super-Noise-Shaped Buttons: [D-D •], [16 •], [DRE "], [DEC •] Leds : 20("), N-S (•), A-D(")	[0000] 0xxx	D-D decode to 16-bit media with super noise-shaper for CD.
MODE 8 UnDre, 16-bit, flat dither Buttons: [D-D •], [16 •], [DRE "], [DEC •] Leds : 20("), N-S ("), A-D(")	[0000] 1xxx	D-D decode to 16-bit media with flat dither; typically to transfer from coded 16-bit to 16-bit PCM with 93dB SNR.
MODE 9 UnDre, 20-bit, NO dither Buttons: [D-D •], [16 "], [DRE "], [DEC •] Leds : 20(•), N-S ("), A-D(")	[0100] 1xxx	D-D decode to 20-bit media with NO dither; typically to transfer from coded 16-bit to 20-bit linear archive or post-production media.
MODE 10 DRE, 16-bit media, Buttons: [D-D •], [16 "], [DRE "], [DEC •] Leds : 20(•), N-S (•), A-D(")	[0x10] xxxx	D-D encode to 16-bit media with DRE encoding SW5 is ignored.
MODE 11 Linear D-D, 20-bit, no dither Buttons: [D-D •], [16 "], [DRE "], [DEC "] Leds : 20(•), N-S ("), A-D(")	[00x1] xxxx	D-D transfer 20-20 or 16-16-bit media with no dither or noise-shaping for format conversion to SDIF-2.

NOTE: Rear panel SW1-4 set power-on defaults ONLY

7. PCB LINK SETTINGS

The following PCB links are set upon leaving the factory:

LK1	-	Left Channel analogue input OV to cable screen, factory fitted
LK2	-	Link factory fitted pins 1-2
LK3	-	Link factory fitted pins 1-2
LK4	-	Link factory fitted pins 3-4
LK5	-	Link factory fitted pins 1-2
LK6	-	Link factory fitted to connect only to LK6.1 and no other pin This is used for diagnostic purposes in other positions
LK7	-	Link factory fitted pins 1-2
LK8	-	Link factory fitted pins 1-2
LK9	-	No link 9
LK10	-	Right Channel analogue input OV to cable screen, factory fitted

7.1. Channel Status link setting

It is possible to set the Co-axial output to either consumer or professional Channel Status format. This is set by Link 4 as follows:

Link 4 (LK4) pins 1-2 linked :	Optical & Co-axial outputs have AES/EBU (professional) Channel Status
Link 4 (LK4) pins 3-4 linked :	Optical & Co-axial outputs have SPDIF (consumer) Channel Status (default)

7.2. Analogue input ground connection

The input cable screen is normally wired to pin 1 of the XLR connector. To link the analogue 0V to pin 1 of the input XLR either link 1 (left channel) or link 10 (right channel) should be fitted.

8. FURTHER INFORMATION

Enquiries about this product should be addressed to :

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