Lindos
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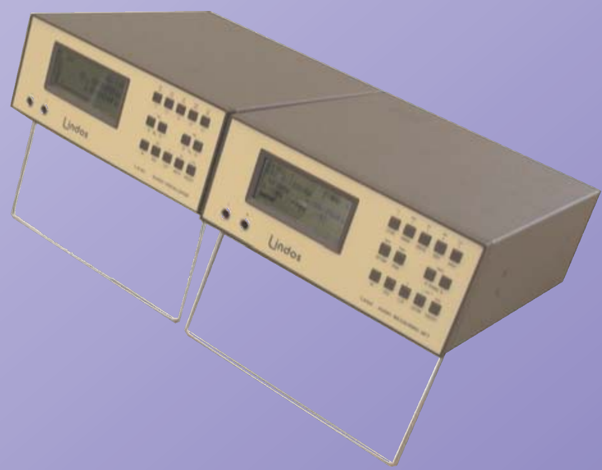


LDM24 Digital Monitoring Adapter

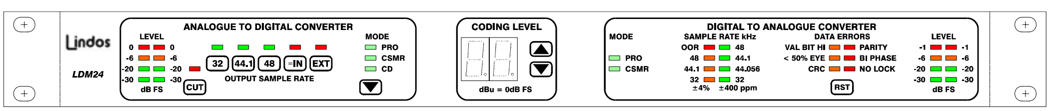
Manual

Issue 1, January 1999

The Lindos LDM24 Digital Monitoring Adapter is a precision 24bit digital to analogue and analogue to digital converter intended primarily as an enhancement for the Lindos LA100 Audio Analyser.

The LDM24 is equally suitable for use as a high quality general purpose converter. In this role it provides a versatile, highly stable unit, with precisely repeatable performance and ease of use.

This manual describes operation of the LDM24 with version 1 control firmware.



Lindos Electronics Saddlemakers Lane, Melton, Woodbridge UK IP12 1PP
Tel +44 (0)1394 380307 Fax +44 (0)1394 385156 email info@lindos.co.uk

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1.0 Getting Started

This section is intended for the first time user, to help in gaining familiarity with the controls. No assumptions as to the previous set-up of the LDM24 have been made. If the instructions are followed precisely then the unit will operate as described, a full explanation of the actions taken will be found later in the manual. If you are using an analogue generator and meter other than the Lindos LA101 and LA102 you will need to interpret these instructions appropriately.

1.1 Setting up

Set up the LDM24 as in Fig. 1, DO NOT switch on the power to the LDM24 just yet.

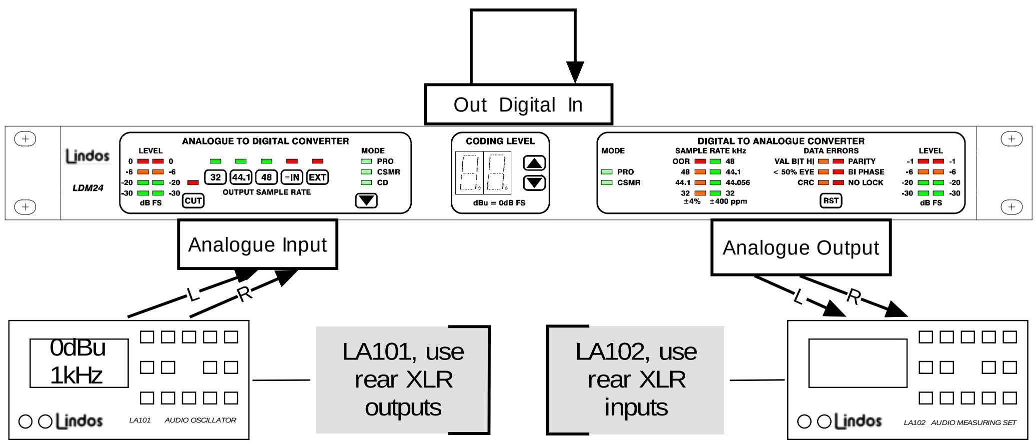

Figure 1 LDM24 and LA100 connections for getting started

NOW switch on the power to the LDM24, while watching its front panel. You will see all the LEDs come on, followed by the sequence of indications shown in figure 2 appearing on the <CODING LEVEL> panel.

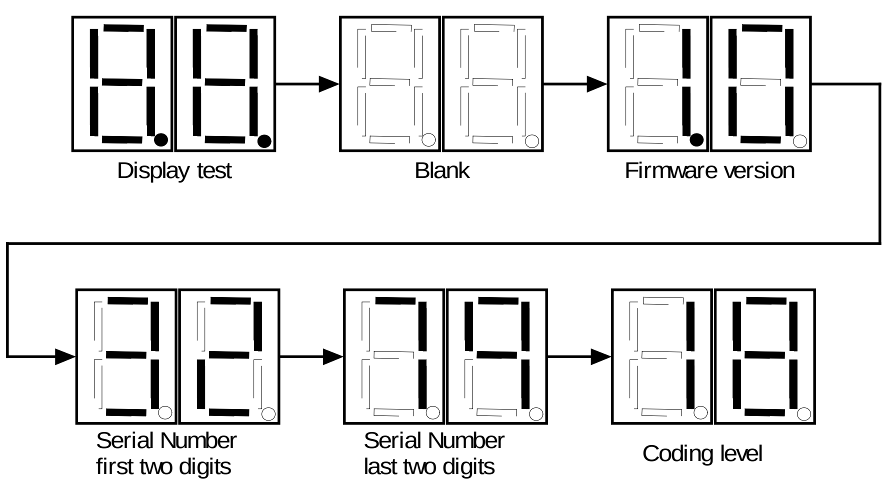

Figure 2 Coding level display sequence at switch on

The display illustrated is for LDM24 serial number 3279, fitted with firmware version 1.0, set to a coding level at which 0dB FS digital gives 18dBu analogue.

1.2 Setting the operating mode

The LDM24 has three user set operating modes, which are described later. For this section you need to ensure that it is in mode 0, as follows:-

Press and hold the "error indication reset" <RST> button, (figure 3), while looking at the coding level panel (figure 4).

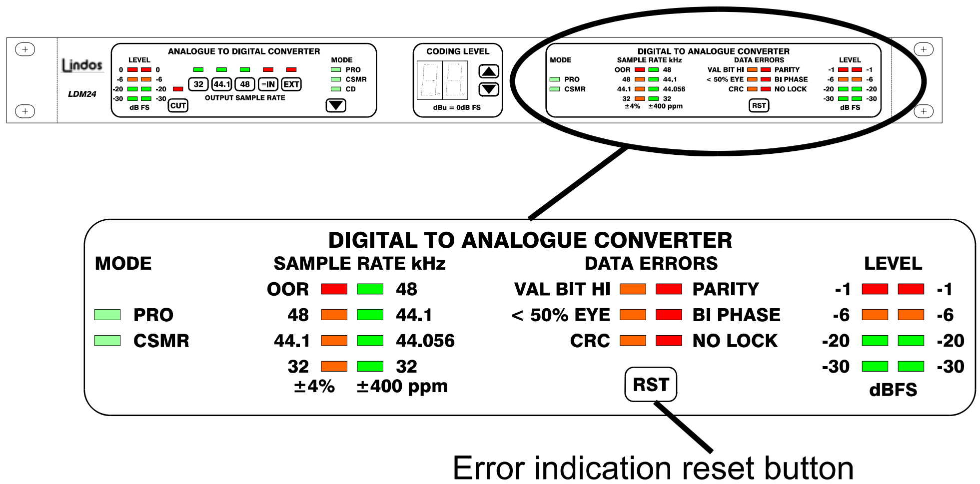

Figure 3 Digital to analogue converter panel

After five seconds the display will change to show **[-][-]**. Release the button. The display will now change to **[P][0]**, **[P][1]**, or **[P][2]**. These indicate user mode 0,1 or 2. The mode can be changed using the "coding level up" and "coding level down" switches shown in figure 4.

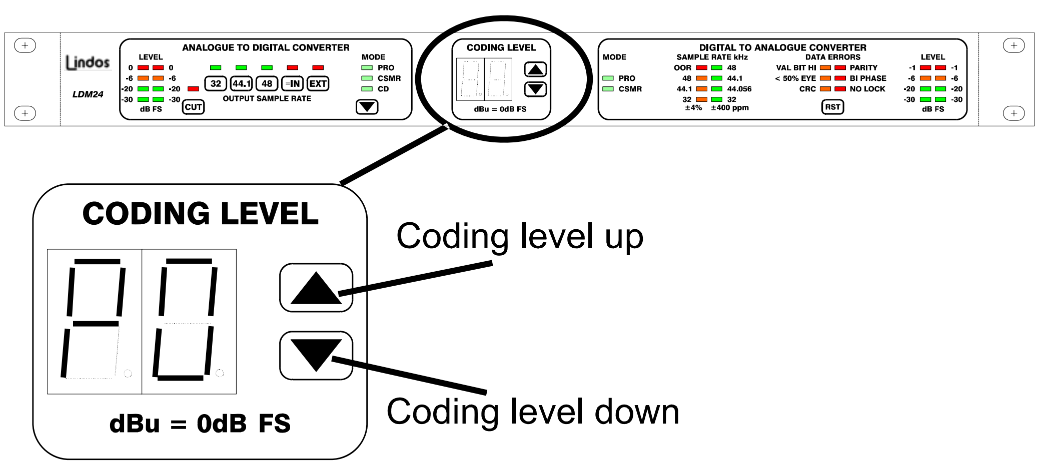

Figure 4 The coding level up and down buttons

Change the selection, up and down, finishing on **[P][0]**. Press and release the <RST> button to store user mode 0. The display will go back to showing coding level.

1.3 Setting the display brightness

Press and hold the <RST> button for five seconds, as in section 1.1, and release the button when the display shows **[-][-]**. The display will change to **[P][0]**. Press and hold the <RST> button again for five seconds, and release the button when the display shows **[-][-]**. The display will change to **[L][L]**.

Press and hold the "coding level up" and then the "coding level down" switches. The display brightness will increase when the up button is pressed and decrease when the down button is pressed.

Choose an appropriate brightness setting, then press and release the <RST> button. The brightness will be stored and the display will go back to showing coding level.

1.4 Audio in and out

Use the coding level up and down switches to set the coding level display to 18.

Refer to figure 5, and press the <CUT> button. The red LED will toggle on and off with alternate presses of the button. Leave it with the LED off. Press the output sample rate 48kHz button. The green LED above it will light.

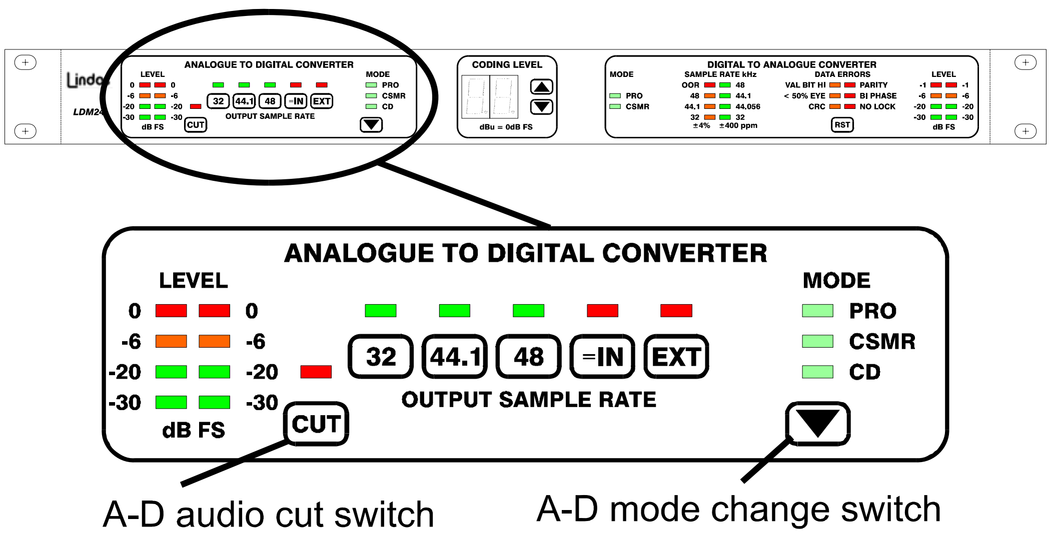

Figure 5 The analogue to digital control panel

With a system connected as in figure 1, and the LA101 output set to 0dBu @ 1 kHz, L+R, the -20 and -30 LEDs on the bargraph displays (extreme left and extreme right) should be lit.

Set the LA102 to twin bar + phase measurement (LEVEL option 4). Both bars should read 0. Push the L/R button on the LA101. The L and R bars should go down in turn, together with the green LEDs on the corresponding LDM24 bargraphs.

Press and hold the <coding level up> button, until the display reads 25. **There should be no change in the readings on the LA102**, the -20 LEDs on the LDM24 should however have switched off. This is because the 0dBu tone from the LA101 was being converted to -18dB FS in the digital signal when the LDM24 was set to 18dBu = 0dB FS, but is now being converted to -25dB FS. The bargraphs on the LDM24 are intended to indicate that a signal is within a sensible working window of digital coding level, they are not intended for programme monitoring. Further explanation of coding level is given in section 4 on page 16.

Change the coding level to 7dBu = 0dB FS, and move it down in single steps. The orange -6 LEDs should come on between 6 and 5, and the red -1 and 0 LEDs at coding levels 1 and 0. This is because the LA101 0dBu tone is now being converted to 0dB FS, the maximum possible level in the digital domain.

If the coding level is now changed to -1dBu = 0dB FS there should be a reduction in the peak level delivered from the digital to analogue converter. This can be best seen using PPM mode (LEVEL Option 5) on the LA102. This is because 0dBu is higher than the level now set for coding 0dB FS - an impossible situation similar to analogue clipping.

Until the overload condition is reached, the reading on the LA102 does not change with coding level. This is because the LDM24 changes the digital to analogue level relationship on its two converters in a complementary manner.

Pressing the <CUT> button on the analogue to digital converter panel (fig. 5) mutes the input to the converter. The action is toggling. Press the button several times to observe its action.

1.5 Digital controls and indications

Look at the digital to analogue converter panel (fig. 3). Several data error LEDs should be lit, as should the sample rate LED for 48kHz ±400ppm.

Push and release the <RST> button. The data errors LEDs should go out. Push the 32kHz button on the analogue to digital converter panel (fig. 5). The sample rate indicated on the digital to analogue converter panel should change, and at least one of the data error LEDs should light.

You have changed the sample rate of the analogue to digital converter, and this has been measured by the digital to analogue converter. The short discontinuity in the digital signal appears as an error condition.

Press the analogue to digital converter "mode change" button several times. The LEDs above it will change between PRO and CSMR. CD will not light. In order to use CD coding mode (the output of the analogue to digital converter will have the format of a CD player) the converter has to be running at a sample rate of 44.1kHz, the same as a CD. *This also applies when the analogue to digital converter is locked to an external digital audio source connected to the digital to analogue converter (<=IN> selected).*

Press the 44.1kHz sample rate button. CD mode can now be selected. If the CD mode is selected with a sample rate other than 44.1kHz then the LDM24 will default to standard consumer coding, but will revert to CD when the sample rate returns to 44.1kHz.

1.6 Connecting other equipment

Connect an item of digital equipment as shown in figure 6.

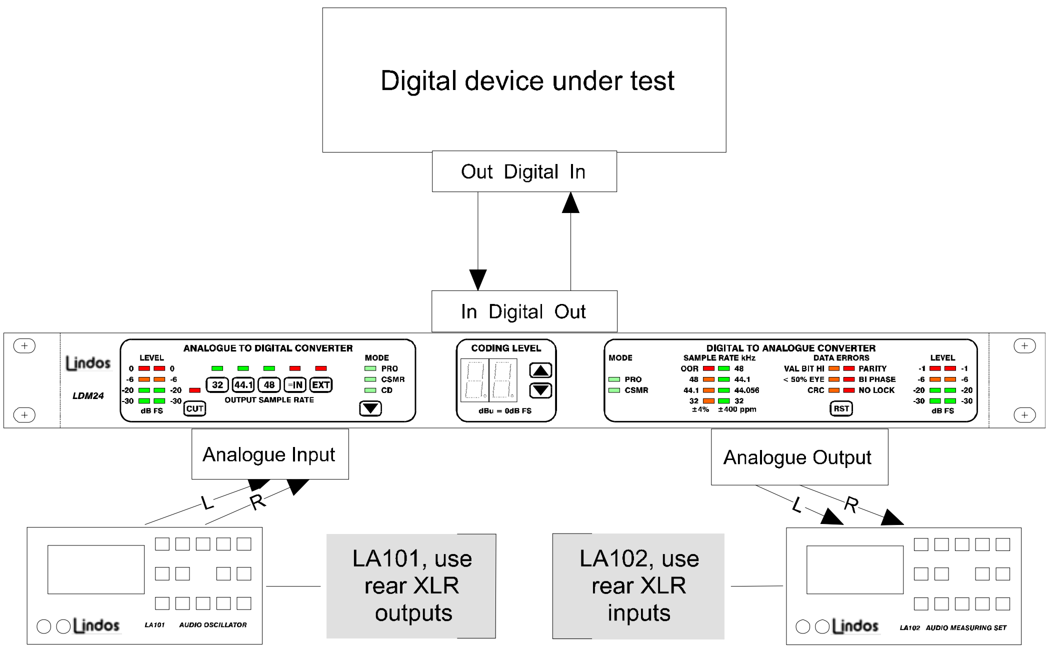

Figure 6 Connecting a device under test

A digital audio recorder is ideal for use in this section.

Generate digital audio from the device connected to the LDM24 digital to analogue converter. Select PPM on the LA102 (level option 5). Switch off LA102 auto-ranging by pressing [*][3] and adjust the range to an appropriate setting, using the RANGE [<][>] keys.

Alter the coding level on the LDM24 using the up and down buttons (fig.4). As the coding relationship increases, the peak level shown on the LA102 will increase.

The peak coding level being produced by the device connected to the LDM24 can be derived.

For:

Peak level displayed on LA102	x dBu
LDM24 coding level setting	y dBu = 0 dB FS
Peak digital coding level	w dB FS

w = x-y

Example:

Peak level displayed on LA102	+8 dBu
LDM24 coding level setting	19 dBu = 0 dB FS

Peak digital coding level = (8-19) = -11 dB FS

Coding level is always zero or negative.

Note:

If the LDM24 is set to 0 dBu = 0dB FS then in the equation above w = x, so the LA102 will in effect read directly in dB FS.

While performing the actions above observe the error and sample rate LEDs on the LDM24, and the indications on the digital to analogue converter panel bargraph display.

It is difficult to demonstrate the effect of coding level change on the analogue to digital converter unless the device connected has some form of input metering. This section assumes that this is present. (If the device is a digital recorder it will typically have metering on which 0dB corresponds to 0dB FS).

Set the LDM24 coding level to 0dBu = 0dB FS
Generate 0dBu tone from the LA101

The indications will now be :

LA101 display reads 0.00dBu	LDM24 bargraphs show 0dB (red)	Digital device shows 0dB

Change the coding level setting to 6dBu = 0dB FS

The indications will now be :

LA101 display reads 0.00dBu	LDM24 bargraphs show -6dB (orange)	Digital device shows -6dB

Change the coding level setting to 20dBu = 0dB FS

The indications will now be :

LA101 display reads 0.00dBu	LDM24 bargraphs show -20dB (green)	Digital device shows -20dB

As the coding relationship increases the digital coding level decreases.

The relationship between levels for the analogue to digital converter is:

Level generated by the LA101	x dBu
LDM24 coding level setting	y dBu = 0 dB FS
Digital coding level	w dB FS

w = x-y

Note:

If the LDM24 is set to 0 dBu = 0dB FS then the LA101 display will in effect read directly in dB FS.

1.7 Summary

If you have worked through the section on getting started you should be familiar with all of the LDM24 controls and indicators.

Detailed information on each of the functions covered can be found in the rest of this manual.

2.0 Analogue to Digital Converter

The analogue to digital converter control panel is shown in figure 5.

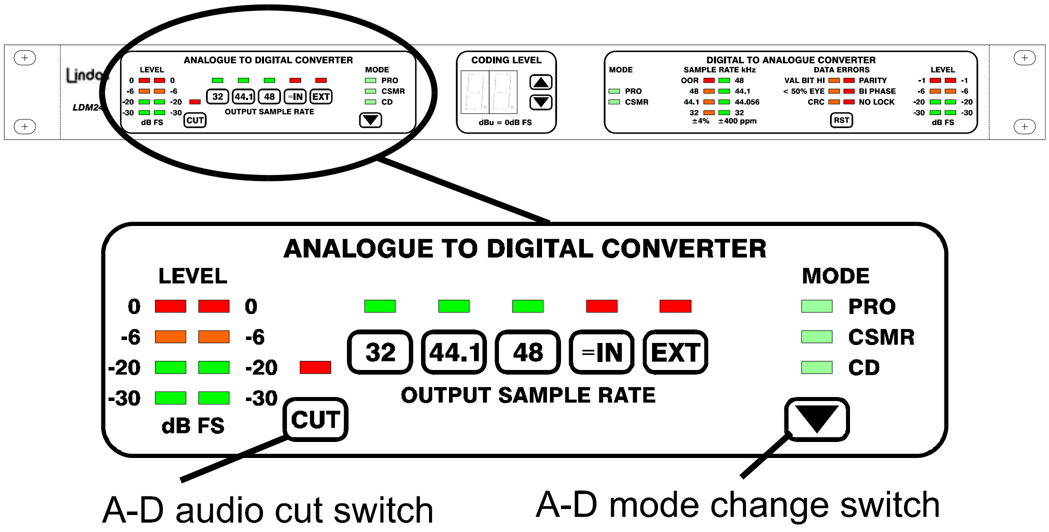

Figure 5 The analogue to digital control panel

The analogue to digital converter accepts balanced stereo audio input on two XLR connectors and provides AES3 digital output, 110 ohm balanced on an XLR connector, 75 ohm unbalanced on a BNC connector.

Sampling rates of 32, 44.1 and 48 kHz are available internally, the converter can be locked to any external AES3 or AES11 source applied to the digital to analogue converter, or an external clock at 256 x sample rate may be applied through a BNC connector on the rear panel.

The converter output can be set to Professional, Consumer or CD mode. This status, together with sample rate information, is carried in the subcode.

2.1 Inputs

The analogue inputs to the LDM24 are electronically balanced and will accept signals up to +25dBu. Maximum digital coding (0dB FS) can be achieved with signals from -6dBu to +25dBu depending on the coding level setting.

Balanced connections are:

Pin 1 Screen (ground)
Pin 2 In phase (+) (hot)
Pin 3 Antiphase (-) (cold)

For unbalanced connection strap pins 1 and 3

2.2 Metering

The stereo bargraph meter is intended to provide an indication that peaks of applied signals are being coded within sensible limits, and that both channels are present. In general, on audio material, one or two of the green LEDs should be on. The indication is of digital coding level NOT analogue input level.

If the red LED comes on, the signal is being coded at the absolute maximum possible level and should be reduced, or alternatively, the coding level setting increased.

Although the set points for the LEDs are precise, these meters are not intended for programme monitoring.

2.3 The CUT function

When the red LED above the button is lit, audio input to the converter is cut. The digital audio bit stream continues to be generated. <CUT> does not generate numerical digital silence, it is an analogue function.

2.4 Sample rates

The standard rates of 32, 44.1 and 48 kHz are generated from high accuracy internal reference oscillators.

When <=in> is selected, the analogue to digital converter is locked to the sample rate and left/right clocks of the bit stream applied to the digital to analogue converter. To use this function, a digital audio bit stream MUST be applied to the digital input of the LDM24. The sample rate of the analogue to digital converter is that indicated on the digital to analogue converter panel.

When <EXT> is selected, an external clock oscillator needs to be applied to the appropriate BNC connector on the back panel. The analogue to digital converter sample rate will be 1/256 of the external clock frequency. As the LDM24 is intended as a stand alone converter, no facility for external Left/Right clock synchronisation has been implemented. The range of sample rates available using an external clock is from 30kHz to 50kHz.

At each change of sample rate selection, the LDM24 resets the internal analogue to digital converter causing a short break in the digital bit stream.

2.5 Analogue to Digital Converter coding modes

Digital audio bit streams contain a subcode which can pass information between items of equipment. The LDM24 analogue to digital converter generates several of these. Control of the format is by use of the mode change button and the selection is indicated on the LEDs above the button.

Table 1:Coding options implemented on the LDM24 analogue to digital converter

Mode: Professional (PRO)

Sample rate selected		Coded as
32kHz		Professional - 32kHz
44.1kHz		Professional - 44.1kHz
48kHz		Professional - 48kHz
=in	**Sample rate detected**	
	32kHz	Professional - 32kHz
	44.056kHz	Professional - 44.1kHz
	44.1kHz	Professional - 44.1kHz
	48kHz	Professional - 48kHz
	Out of range	Professional - Undefined
EXT		Professional - Undefined

Mode: Consumer (CSMR)

Sample rate selected		Coded as
32kHz		Consumer - 32kHz
44.1kHz		Consumer - 44.1kHz
48kHz		Consumer - 48kHz
=in	**Sample rate detected**	
	32kHz	Consumer - 32kHz
	44.056kHz	Consumer - 44.1kHz
	44.1kHz	Consumer - 44.1kHz
	48kHz	Consumer - 48kHz
	Out of range	Consumer - 44.1kHz
EXT		Consumer - 44.1kHz

Note that 44.1kHz is used as the consumer default when rate is undefined

Mode: CD

Sample rate selected		Coded as	Lamp
32kHz		Consumer - 32kHz	CSMR
44.1kHz		CD	CD
48kHz		Consumer - 48kHz	CSMR
=in	**Sample rate detected**		
	32kHz	Consumer - 32kHz	CSMR
	44.056kHz	CD	CD
	44.1kHz	CD	CD
	48kHz	Consumer - 48kHz	CSMR
	Out of range	Consumer - 44.1kHz	CSMR
EXT		CD	CD

Note that 44.1kHz is used as the consumer default when rate is undefined

The LDM24 encodes the type of audio and the sampling rate. See Table 1 above.

CD mode is a special case of consumer mode. It will only be encoded by the LDM24 when external clock or 44.1kHz sample rate are selected. If CD mode is selected when neither of these conditions are met, then the LDM24 will default to consumer mode, which will also be indicated on the analogue to digital converter control panel.

Note that CD mode is always permitted with an external 256 x sampling rate clock, which allows out of tolerance CD coded signals to be generated for testing of digital to analogue conversion equipment.

Do not confuse the consumer mode with the widely used S/PDIF interface found on consumer products. S/PDIF refers to the Sony/Philips digital interface format, which describes voltage levels and impedance for use at the connector, rather than the content of the bit stream.

When using the LDM24 with consumer digital audio equipment we recommend the use of an XLR to S/PDIF adapter rather than using the unbalanced AES output on the BNC connector, as the signal voltages are different.

3.0 Digital to Analogue Converter

The digital to analogue converter panel is shown in figure 3

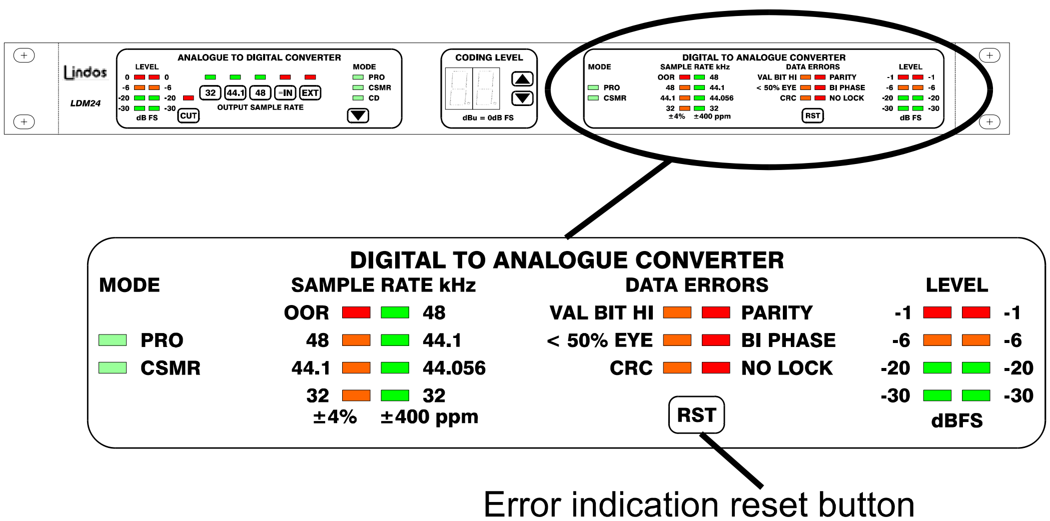

Figure 3 Digital to analogue converter panel

The digital to analogue converter accepts AES3 digital input, 110 Ω balanced on an XLR connector, 75 Ω unbalanced on a BNC connector and provides balanced stereo audio output on two XLR connectors.

As both balanced and unbalanced inputs are always active only one of the inputs should be connected at a time.

The converter will accept digital audio with sampling rates between 30kHz and 50kHz. It can be used as the timing source for the analogue to digital converter.

3.1 Outputs

The analogue outputs from the LDM24 are electronically balanced, 50 Ω and will deliver signals at up to 25dBu into 10 kΩ.

Balanced connections are:

Pin 1 Screen (ground)
Pin 2 In phase (+) (hot)
Pin 3 Antiphase (-) (cold)

For unbalanced connection use pin 2 for live, pin3 for screen. The screen can be strapped to pin 1 at the LDM24 if the screen connection on the equipment connected to it is not grounded.

3.2 Metering

The stereo bargraph meter is intended to provide an indication that peaks of signals in the applied digital audio bit stream have been coded within sensible limits, and that both channels are present. In general, on audio material, one or two of the green LEDs should be on. The indication is of digital coding level NOT analogue output level.

If the red LED comes on, the signal has been coded within 1dB of the absolute maximum possible level. This condition will sometimes be encountered, particularly on CDs. It should generally be investigated as digital audio systems have **NO** headroom above 0dB FS.

Although the set points for the LEDs are precise, these meters are not intended for programme monitoring.

3.3 Data errors

The LDM24 checks every 8 milliseconds for the worst data error encountered in the applied digital audio bit stream. This information is then stored and displayed. The errors found are aggregated on the display, which can be reset with the <RST> button.

Note that this is the only control on the LDM24 which acts when it is released. *Do not hold the button pressed for more than 4½ seconds as this will activate other functions*

The errors detected and displayed are:

Validity bit high.
Indicates that the validity flag for a sample was high, indicating non audio data.

< 50% eye.
The eye width has been less than half a bit period, indicating noise or other interference on the data.

CRC (Professional mode only).
An incorrect cyclic redundancy check code has been detected. This can be caused by incorrect re-assembly of blocks after editing

Parity error.
Parity calculated on received data does not agree with parity bit encoded in data

Biphase coding error.
The received bit stream has violated the rules of biphase coding used for digital audio

No lock to incoming data.
Usually means no bit stream is present

Note that the last three errors will often result in loss of audio.

3.4 Sample rates

The sample rate of the received bit stream is measured with reference to one of the LDM24 internal references. The frequency and accuracy are displayed.

OOR (out of range) means that the sample rate is outside the range of the seven reported rates. It DOES NOT mean that no digital audio is present unless the NO LOCK error is also present.

Note that if the analogue to digital converter is using clocks from the digital to analogue converter (=in condition) then CD coding is only allowed when one of the three rates corresponding to 44.1kHz is detected. (Please see Table 1 on Page 12)

3.5 Digital to analogue converter mode indication

The basic consumer (CSMR) and professional (PRO) information from the subcode is displayed.

4.0 Coding Level

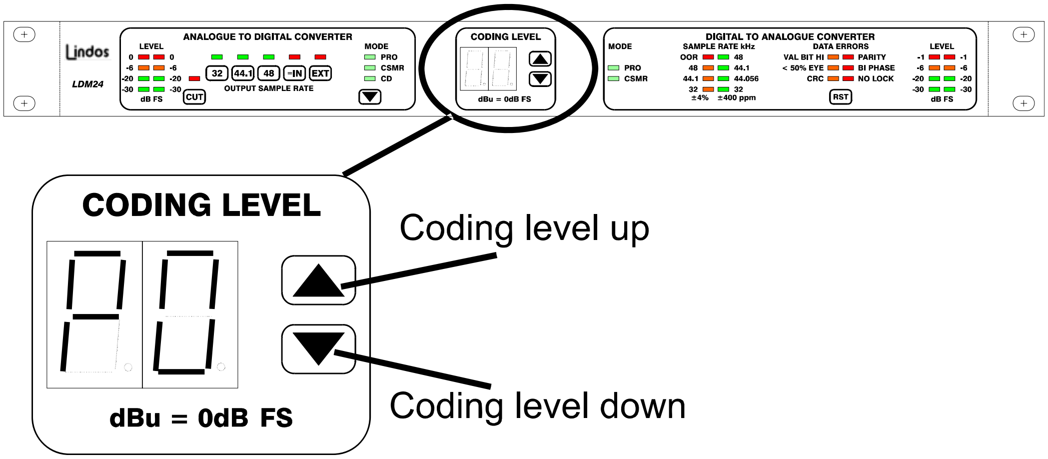

Figure 4 The coding level up and down buttons

The controls and display for coding level are shown in figure 4.

The relationship between signal voltage in the analogue domain and coding in the digital domain on the LDM24 is precise, and can be varied in 1dB steps over a 32dB range.

When the LDM24 coding level setting is changed, the analogue to digital converter and digital to analogue converter gains are adjusted simultaneously.

4.1 The meaning of coding level in digital audio

In a digital system, the largest and smallest numbers which can be represented are well defined. The largest is when every binary digit is a 1, the smallest when all are 0. Digital audio is coded using the twos complement number system to represent positive and negative instantaneous signal amplitudes so the actual numbers are not quite that simple, but the principle still applies, that there is a highest and lowest number that can be represented. A signal whose maximum peaks reach the maximum number and negative peaks reach the minimum number is referred to as Full Scale.

Coding level is the term used to describe the ratio of the actual signal amplitude to the maximum possible signal amplitude that can exist in the system. It is normally expressed in decibels. This gives a scale with 0dB as the maximum and all other values as a negative number of dB. To indicate that the value is relative to the digital Full Scale Signal it is written as dB FS

4.2 The relationship between analogue and digital signal level

In analogue audio there is a reference level, and signal headroom above this. (German and Scandinavian practice is to define the maximum signal and work down, which requires a slight change of analysis)

When testing equipment or transmission systems with steady, defined, signals, it is easy to define the largest signal within the system. When audio material is present the situation changes.

Where VU monitoring is used (with the meter reading dependent on audio power), the peak amplitude present for a particular meter reading is dependent on the ratio of peak to mean signal, which is waveform dependent. VU metering is generally set so that reference 1kHz sinewave tone at +4dBu gives a reading of 0VU. Signal peaks can be typically over 20dB higher than this reference.

For analogue Peak Programme Meters (PPMs), the reading is still not a true peak signal value. The reading is quasi peak, and the characteristics were optimised for analogue systems. This means that short duration transient signals will read less than their true peak value. How much less depends on the integration time of the meter, and the nature of the programme material. With the BBC PPM, true signal peaks can typically be up to some 10dB higher than indicated, depending on programme content.

When analogue signals are converted to digital signals, then the digital domain has to be treated in the same way as an analogue amplifier with hard clipping. Sufficient room has to be left for the peaks of signals, and the relationship between analogue and digital signals needs to be defined.

Two common relationships are in use. In North America and other areas using VU metering, a ratio of 25dB between analogue reference and digital full scale is used. The EBU recommend 18dB for Europe, when PPM metering is used.

In the context of a line level reference of 0dBu (0.775v r.m.s.) these are expressed as 25dBu = 0dB FS and 18dBu = 0dB FS.

The LDM24 allows precision variation between -6 and 25dBu = 0dB FS

Many different metering characteristics and scalings are in use, and we have been unable to trace precise definitions for all of them. Figure 7 is based on the best information we have been able to obtain. The relationships between a signal in the analogue and digital domains at three relative coding levels is shown.

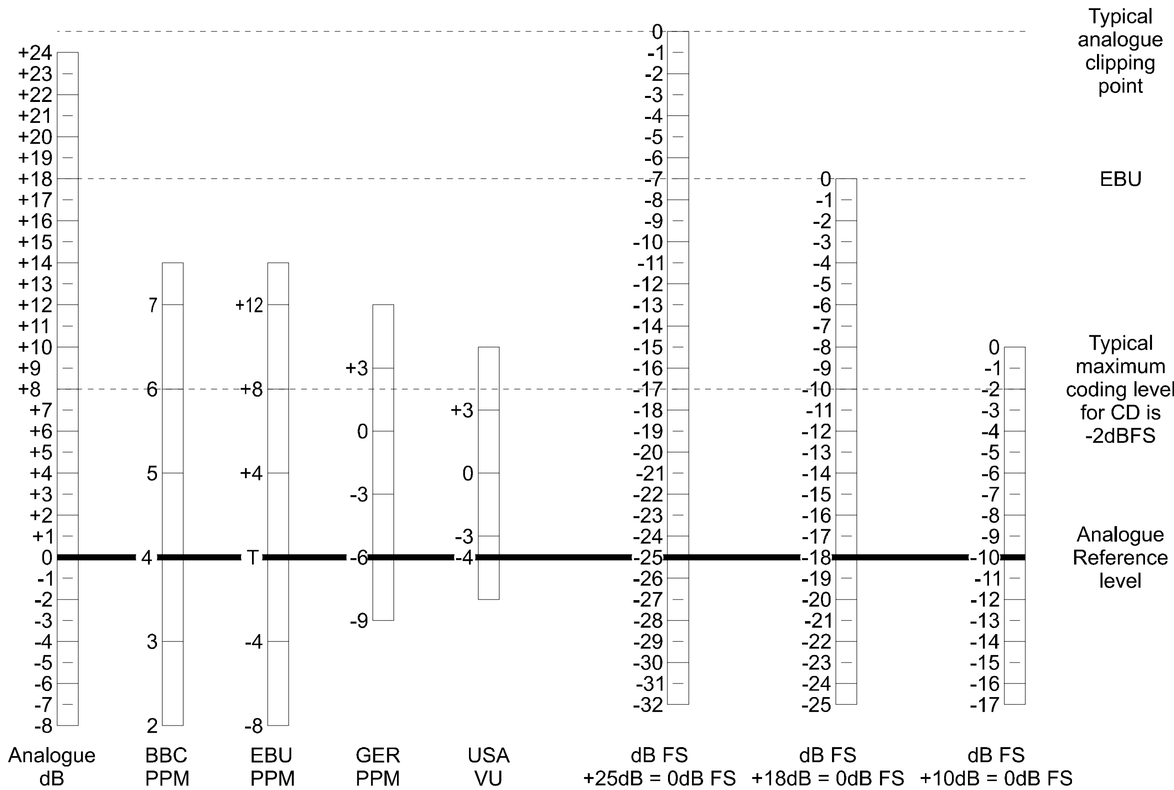

Figure 7. Analogue metering and scaling comparison

4.3 Setting the LDM24 for test signal use

Once the range of coding levels is known, and from this the reference coding level is derived, the relationship between dBu and dB FS for an LA100 connected to an LDM24 can be calculated. This is best illustrated by example.

If a digital radio link is to be used with 20dB headroom above analogue reference, and analogue reference is 0dBu, then an analogue signal applied at 0dBu will be coded 20dB below the maximum possible digital signal level. If an analogue signal is present at +20dBu this will be coded at the maximum possible digital signal level, or 0dB FS. Setting the LDM24 coding level control to 20dBu = 0dB FS will give this condition.

If an LDM24 is connected at each end of the link as shown in figure 8a with an LA101 and an LA102 connected to it as shown, any analogue test required on the link can be performed using the LA100 as if it were connected to an analogue link as shown in figure 8b.

NOTE If the coding level controls at both ends are changed but still remain the same as each other, then levels measured on the LA100 will still be the same, only the digital link headroom will have changed.

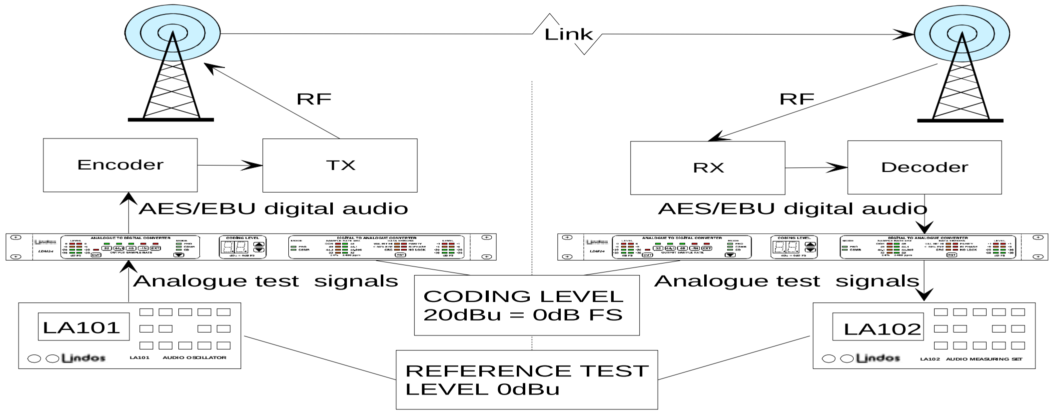

Figure 8a Test set-up for analogue fed digital radio link

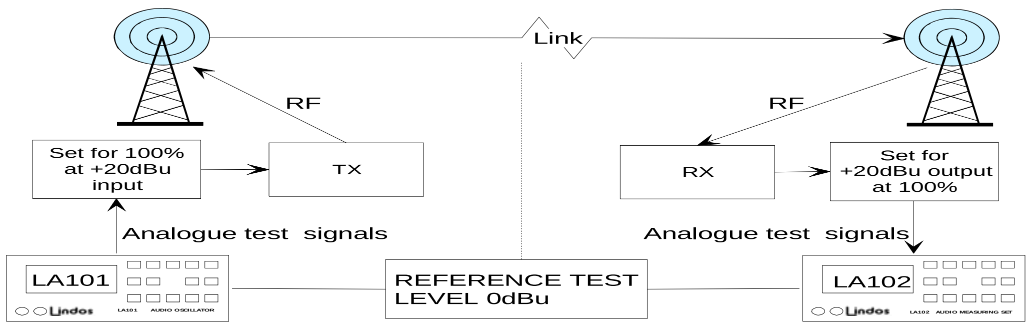

Figure 8b Test set-up for equivalent analogue radio link

4.4 Setting the LDM24 for use with audio material

This section describes procedures for two extremes in each domain. These are live type material where little or nothing has been done to control dynamics or overall system gain, and material which has been controlled and where the characteristics are known. In practice there is no substitute for experience.

4.4.1 Digital audio material with little control of level

Owing to the low noise floor of 24bit digital audio, it is possible to use it without any control other than basic gain setting in applications as diverse as recording classical piano and providing audio feeds of conference stage microphones.

The bargraph meter on the LDM24 digital to analogue converter panel should help indicate the magnitude of the peaks in such material.

It is important when setting the operating point of a converter to avoid distortion in the analogue system being driven. If the analogue system can accept an input of +25dBu or higher, then there is no problem. No digital audio signal can exceed 0dB FS, so the converter can be set to 25dBu = 0dB FS. If the maximum input to the analogue device is lower, then a setting a few dB less than the maximum is safe.

The exception is when an analogue device with a poor noise performance is to be connected, and the coding level of the digital audio to be converted is consistently low. It may under these conditions be necessary to increase the analogue signal by increasing the coding level setting and accepting the risk of distortion.

4.4.2 Digital audio material with controlled level

In this case either the highest coding level is known, or the line-up level of the system which encoded the audio is known.

For the case where the highest coding level is known, and the highest signal level which can exist in the analogue domain is also known, the required setting can be calculated.

Example 1:

A digital hard disc recording needs to be converted to EBU analogue levels to be fed into a loudspeaker and PPM monitoring unit. The material on the hard disk has been computer normalised to 2dB below peak coding (-2dB FS)

Calculation:

The peak signal at the input to the analogue monitor needs to be +8dBu. This must be equivalent to -2dB FS. 0dB FS would therefore correspond to +10dBu.

The LDM24 needs to be set to 10 on the coding level display.

For the case where the line up level of the system which encoded the audio is known, the calculation is similar.

Example 2:

A broadcaster following EBU standards, where peak signal is 8dB above analogue reference, instructs users of digital recorders to set 0dB analogue reference tone to be 20dB below peak digital coding level.

The digital signal from this needs to be fed into the same monitoring unit as in example 1.

Calculation:

Peak signals in the original material would have been 8 dB above reference, which was set to -20dB FS. The peak digital coding, ignoring any overshoot, will therefore be (-20 + 8) = -12dB FS. This level signal needs to appear at the input to the monitoring system at +8dBu. Since +8dBu must be equivalent to -12dB FS, +20dBu needs to be equivalent to 0dBFS

The LDM24 needs to be set to 20 on the coding level display.

4.4.3 Analogue audio material with little control of level

Great care is needed when converting audio of this type to the digital domain.

The problem is analogous to that encountered when passing audio through an analogue amplifier with a well defined clipping point. It is essential that peaks do not exceed the maximum for the system, in this case 0dB FS, while ensuring that the lower levels are not too low, leading to poor signal to noise and in this case unnecessarily few bits per sample.

If accurate programme metering equipment is available this should be used, and an appropriate coding level calculated (see 4.4.4 below). If not, then the bargraph meters on the LDM24 can be used as a guide.

With analogue audio applied to the analogue to digital converter inputs, wait for a loud part of the material and adjust the coding level so that the -6dB FS light flashes on occasionally. Watch the display and ensure that the red 0dB FS LED does not light. If it does, re-adjust the coding level. If the -30dB FS LED stays unlit for long periods the level is probably too low.

4.4.4 Analogue audio material with controlled level

Much analogue audio has been controlled during production, using Peak Programme or VU meters to monitor levels. In both cases peaks will normally be present which are greater than the nominal maximum level. This is because PPMs are quasi-peak devices (they do not respond fully to signals of short duration) and VU meters are intended to indicate the volume which will be perceived by the listener, so the relationship to peak signal level depends on the waveform of the audio being monitored.

Even when an analogue limiter has been used, there will normally be overshoots where high level transients have occurred.

The bargraph meters of the LDM24 indicate peak level, and have a sufficiently long dwell time for very short transients to be seen. This helps in determining the appropriate coding level set point.

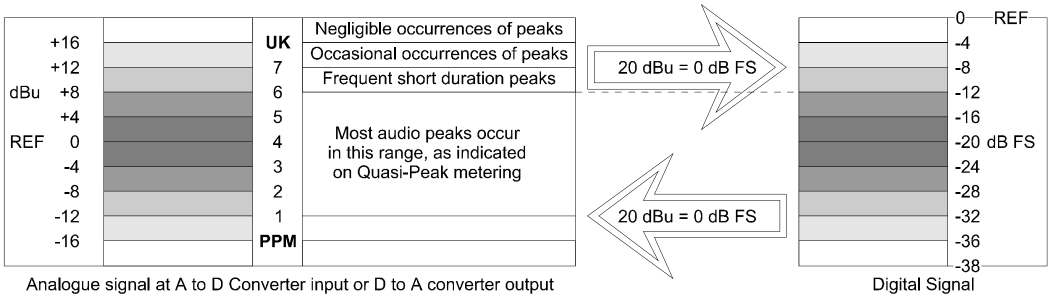

Figure 9a: Typical Digital and Analogue signal relatonships for controlled audio

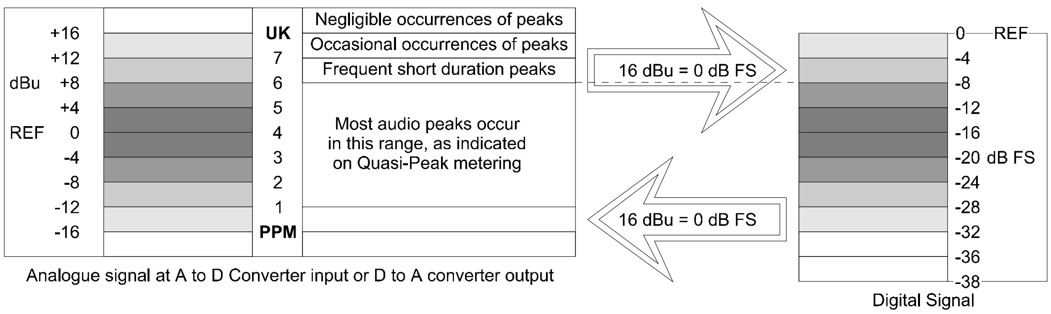

Figure 9b: As Figure 9a with less digital headroom

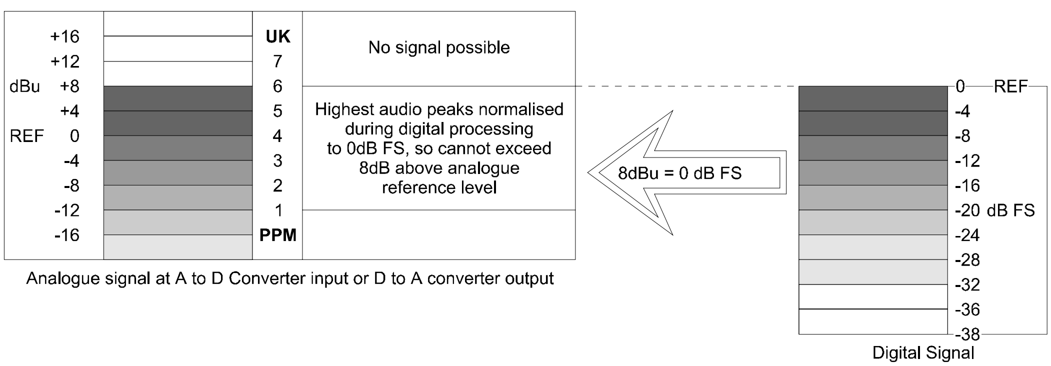

Figure 9c: Relationships for a Digital source with peak output normalised to 0dB FS

When the highest signal level which should exist in the analogue domain is known, the nominal required setting can be calculated.

Example 1:

A reel to reel recording needs to be converted to digital. Analogue levels measured on a PPM monitoring unit show Left and Right containing peaks to +8dB above reference level. The system output has an output of 0dBu at 0dB reference level. 4dB headroom is to be allowed to account for peaks above those shown on the analogue meter.

Calculation:

The maximum peak signal being allowed for at the analogue input to the LDM24 is +8dB (measured) + 4dB headroom = +12dB. This signal is to correspond to 0dB FS.

The LDM24 therefore needs to be set to 12 on the coding level display.

Example 2:

A broadcaster following EBU standards, where analogue peak signal is 8dB above analogue reference, instructs users of digital recorders to set 0dB analogue reference tone to be 20dB below peak digital coding level.

0dBu needs to be equivalent to -20dB FS, so +20dBu needs to be equivalent to 0dB FS

The LDM24 needs to be set to 20 on the coding level display.

5.0 Operating Modes

The LDM24 can operate in one of three modes.

In mode P0 the LDM24 will start up with all adjustable settings (Analogue to Digital Converter Cut, Sample Rate, and Mode, and Coding Level) as they were when the unit was last switched off. These can all then be re-adjusted normally from the control panel.

In mode P1 the LDM24 will start up with all adjustable settings (Analogue to Digital Converter Cut, Sample Rate, and Mode, and Coding Level) in a preset condition. These can all then be re-adjusted normally from the control panel.

In mode P2 the LDM24 will start up with all settings (Analogue to Digital Converter Cut, Sample Rate, and Mode, and Coding Level) fixed. Apart from the functions of the <RST> button all front panel controls are inoperative.

5.1 Selecting an operating mode

Press and hold the <RST> button on the digital to analogue converter panel. After five seconds the coding level display will show **[-][-]**. Release the <RST> button. The display will show **[P][0]**, **[P][1]**, or **[P][2].**

Switch between these using the Coding Level up and down buttons. Selection is made by pressing the <RST> button for a normal, short, period. (The LDM24 will then resume normal operation).

5.2 Setting the preset condition for modes 1 and 2

First set the LDM24 to mode P0 and set up the unit in the condition required.

Next select mode P1 or P2 as required, and press the <RST> button. The settings of the LDM24 will be stored in non volatile memory.

5.3 Situations where a particular mode is recommended

When the LDM24 is installed in a situation where loudspeakers, or other equipment which may experience problems with high audio levels, are connected to it, we recommend the use of mode P1, so that at switch-on the conversion gain, and hence audio levels, are defined.

Where the LDM24 is installed as part of a monitoring system and where non-technical staff have access to equipment, we recommend that mode P1 or P2 is selected before the equipment is left. This ensures that settings cannot accidentally be altered or can be easily restored.

Great care should be taken to ensure the presence of a reference if the LDM24 is set to power up in either the EXT or =in sampling condition. This possibility can be avoided using P1.

6.0 Use as a Studio Converter

The LDM24 can be used as a high quality studio grade converter. Great care has been taken to ensure that both converters will provide excellent audio quality in addition to precise conversion for instrumentation use.

Please note however that no muting is provided on detection of faulty or discontinuous bit streams. ***We recommend that monitoring loudspeakers are dimmed or cut before any changes are made to LDM24 settings, or until a stable bit stream is being received.***

The LDM24 has the advantage over many converters made for studio use that all settings are precisely repeatable, with no tracking errors.

When using the LDM24 as a converter for audio material we recommend that it is set to user mode 1, preferably with the analogue to digital converter panel set to CUT to ensure that it does not power-up in a condition which will generate excessively high sound levels on monitoring loudspeakers.

7.0 Technical Details

7.1 Technical Description

The LDM24 consists of three main electronic sub assemblies. These are inter-connected via cable looms to allow easy replacement of any sub-assembly. The schematic diagrams represent the main circuit board assembly including the main rear panel connectors. These diagrams are for **reference only** and are not maintained by Lindos Electronics.

- **Mains Power Supply**

The power supply assembly consists of the mains IEC320 connector with two integral 20mm 2A (M)edium Speed fuses and EMC filter. This is connected directly to the Switch Mode Power supply that is used to provide the three DC supplies for the main circuit board. A connector allows the IEC connector to be removed from the power supply.

The connections to the circuit board are made via an integral loom and a connector allows removal of the power supply from the circuit board loom.

The DC supplies are :-

+5VD	Digital Supply	Red
DGND	Digital Ground	Black
+15V	Analogue Supply	Yellow
AGND	Analogue Ground	Black
-15V	Analogue Supply	Grey

The power supply contains no serviceable components and in event of a failure should be replaced as an assembly. Please contact Lindos Electronics for replacement parts.

- **Front Panel Assembly**

The front panel assembly consists of the membrane switch matrix, aluminium front plate and a circuit board. The circuit board contains all the front panel LEDs, displays and driver IC. A single ten way flexi-circuit interfaces the membrane switches to the front panel circuit board. A 16 way IDC ribbon cable interfaces the main circuit board with the front panel assembly. The front panel assembly is attached to the LDM24 chassis with six pan head screws, these are accessible from inside the chassis.
The front panel circuit board contains no adjustable components and therefore should not be removed from the front plate.

NOTE To remove the main circuit board from the chassis it is necessary to remove the flexi-circuit from the front panel circuit board header strip connector.

- **Main Circuit Board PCBA-2400-0000**

The main board contains all the LDM24 digital and analogue circuitry. As this unit is factory calibrated it is **strongly recommended** that none of the potentiometers or trimmer capacitors are adjusted.
If the unit requires re-calibration it is recommended that the unit is returned to Lindos Electronics.

The table below defines the function of each of these components :-

Adjustment	Function
RV1	Right Channel Gain Trim
RV2	Right Channel Input Balance Trim
RV3	Left Channel Gain Trim
RV4	Left Channel Input Balance Trim
RV5	Right Channel Output Balance Trim
RV6	Right Channel Output Gain Trim
RV7	Left Channel Output Balance Trim
RV8	Left Channel Output Gain Trim
CV1	Right Channel H.F Input Balance
CV2	Left Channel H.F Input Balance

The circuit board also contains links to cater for different grounding techniques on all of the input and output connectors.

Due to EMC considerations it is strongly recommended that these are not changed. Compliance with current CE emission limits is only guaranteed if the grounding structure is not modified from the original configuration.

In some cases these links are replaced by braid to provide improved grounding of the connector pins to the LDM24 chassis. If required these braid links must be removed in conjunction with the circuit board links.

Link	Connector / Pin	Signal	Function
1	CONN1/1	AES Digital In - Screen	Pin 1 to LDM24 Chassis
3	J1	SPDIF Digital In - Screen	Coax Screen to LDM24 Chassis
2	CONN2/1	AES Digital Out - Screen	Pin 1 to LDM24 Chassis
4	J2	SPDIF Digital Out - Screen	Coax Screen to LDM24 Chassis
5	J3	Ext. Sample Rate - Screen	Coax Screen to LDM24 Chassis
6	J3	Ext. Sample Rate - Screen	Coax Screen to LDM24 Digital Ground
8	CONN3/1	Right Analogue In - Screen	Pin 1 to LDM24 Chassis
11	CONN4/1	Left Analogue In - Screen	Pin 1 to LDM24 Chassis
13	CONN5/1	Right Analogue Out - Screen	Pin 1 to LDM24 Chassis
16	CONN6/1	Left Analogue Out - Screen	Pin 1 to LDM24 Chassis

7.2 Specifications

Analogue-Digital Section

Sample rates supported	32kHz, 44.1kHz, 48kHz, External, Locked to incoming.
Sample rate frequency accuracy	±25ppm (0.0025%)
Conversion accuracy	±0.1dB at or above 0dBu=0dB FS, < 0dBu=0dB FS, ±0.2dB
Frequency response	fs=32kHz ± 0.1dB 20Hz-14kHz fs=44.1kHz ± 0.1dB 20Hz-20kHz fs=48kHz ± 0.1dB 20Hz-21kHz
Noise and distortion at +25dBu=0dB FS coding level	THD+N <-80dB (0.01%)
Noise, RMS unweighted, 22-22kHz bandwidth	<-92dB below peak coding
Muting	Digital silence, to the limits of the A-D converter

Digital-Analogue Section

Sample rates supported	32kHz, 44.056kHz, 44.1kHz, 48kHz, External
Sample rate frequency accuracy indication	±4%, ±400ppm (0.04%)
Conversion accuracy	±0.1dB at or above 0dBu=0dB FS, < 0dBu=0dB FS, ±0.2dB
Frequency response	fs=32kHz ± 0.1dB 20Hz-14kHz fs=44.1kHz ± 0.1dB 20Hz-20kHz fs=48kHz ± 0.1dB 20Hz-21kHz
Noise and distortion At +25dBu=0dB FS coding level,	THD+N <-80dB (0.01%)
Noise, RMS unweighted, 22-22kHz bandwidth	<-100dB below peak coding
Power supply	85-264 volts 47-440Hz, 3 pin IEC connector
Dimensions	483 mm (19”) rack mount, 44mm (1 ¾”- 1U) high Optionally, the LDM24 is available without rack mount ears for bench use.

7.3 Certificate of Conformity

CERTIFICATE OF CONFORMITY

Manufacturers name	**Lindos Electronics**
Address	**Saddlemakers Lane Melton, Woodbridge Suffolk IP12 1PP U.K.**
Telephone	**+44 (0) 1394 380307**
Fax	**+44 (0) 1394 385156**
email	**info@lindos.co.uk**
Product name	**Lindos Digital Monitoring Adapter**
Model number	**LDM24**

Complies with the requirements of the European Low Voltage Directive and Electromagnetic Compatibility Regulations

and

conforms to the following standards:-

Safety:	**BS EN 61010-1:1993 and AMD 8691**
EMC:	**BS EN 50081-1: 1992 (Emissions) BS EN 50082-1: 1998 (Immunity) BS EN 60555-2: 1987 (Mains harmonics)**

Chris M. Skirrow
Proprietor

Lindos
Electronics

Lindos Electronics
Saddlemakers Lane,
Melton, Woodbridge,
Suffolk IP12 1PP UK

Tel: +44 (0) 1394 380307
Fax: +44 (0) 1394 385156
email: support@lindos.co.uk