LM1 MONITOR & LG1 GENERATOR
USER MANUAL

1.0 LM1 MONITOR

2.0 LG1 GENERATOR

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LM1
PORTABLE DIGITAL / ANALOGUE AUDIO MONITOR

1.1 GENERAL DESCRIPTION

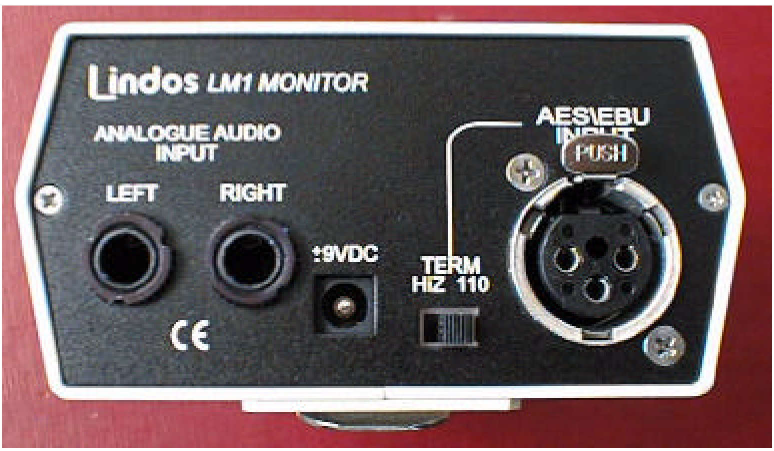
The **LM1** is a low cost, portable beltpack unit for monitoring AES/EBU digital or analogue audio signals. The **LM1** has a headphone output and LEDs to display basic signal parameters and the integrity of a digital audio signal. It is a companion to the Lindos **LG1** portable digital / analogue audio generator, and will operate with the Lindos **LA100** and **LA100-D** Audio Analyser, or any other audio source.

The **LM1** is well suited for use in professional and consumer environments such as broadcast installations, production studios, signal distribution facilities or any other place where digital or analogue audio signals are used. Analogue or digital programme audio can be listened to on the stereo headset output, whilst left and right levels are displayed on the LED bargraph meters.

When monitoring AES/EBU signals, the **LM1** displays many critical parameters such as sampling frequency, audio level, emphasis, professional or consumer sub-code and any data error that may be occurring. The digital input will accept signals with sampling frequencies from 30kHz to 50kHz.

1.2 OPERATIONAL FEATURES

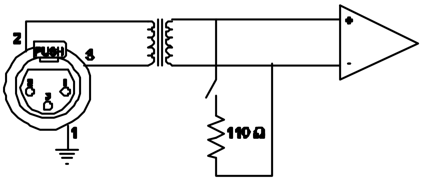
FRONT PANEL



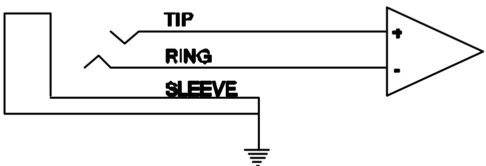
REAR PANEL

1.3 CONNECTING THE LM1

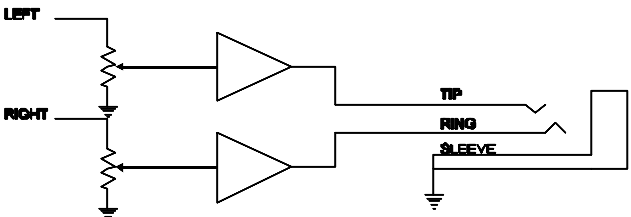
The **LM1** uses a 3-pin XLR for an input connector and is wired as shown below. The input impedance can be switched between 110 Ω and “HiZ” (>20kΩ). If unbalanced 75 Ω monitoring is required, we recommend the use of a 75 Ω to 110 Ω impedance matching transformer.



BALANCED DIGITAL AUDIO INPUT



BALANCED ANALOGUE AUDIO INPUT



HEADPHONES OUTPUT

1.4 DISPLAYS

The **LM1** is equipped with a stereo bargraph audio meter. Two columns of twelve LEDs cover the range from -39 to 0 dB FS digital, and -21 to +18 dBu analogue.

The following STATUS indicators may be illuminated if the digital audio bit stream is in good condition:

AUDIO -	Data detected is true AES/EBU audio data
PRE-EMPH -	Encoded channel emphasis is enabled
CONSUMER -	The channel status data block is set for consumer use
LOCK -	Correctly locked to a signal, but with a sampling frequency other than 32 kHz ,44.1 kHz or 48 kHz
32 kHz -	Locked to a signal with a sampling rate of 32 kHz
44.1 kHz -	Locked to a signal with a sampling rate of 44.1 kHz
48 kHz -	Locked to a signal with a sampling rate of 48 kHz

If the STATUS/ERROR switch is in the ERROR position, the NO ERROR LED will be illuminated. It is recommended that the switch be left in the STATUS mode unless the RCV ERROR LED is illuminated.

The RCV ERROR indicator will illuminate if the digital audio signal is absent or the data is corrupted. Line receive errors are prioritised and only the higher priority error will be indicated.

NO LOCK -	Signal absent or data corrupted beyond recoverable limits (highest priority)
BIPHASE ERROR -	Bi-phase coding of the signal is incorrect
PARITY ERROR -	Parity bit of the incoming data is not set as specified
CRC ERROR -	CRC (Cyclic redundancy check) value calculated for the received signal does not match the CRC byte of the channel status word.
CONFIDENCE FLAG-	Indicates an eye pattern violation. It is a warning that the incoming signal is approaching the limits beyond which reliable recovery may be jeopardised. This is typically caused by the band limiting effects of long cable runs.
VALIDITY BIT- V	Validity bit of AES/EBU bit stream is set high to

indicate that the signal is not suitable for D to A conversion

In analogue mode, the RCV ERROR LED indicates power on.

POWER

The **LM1** is powered by four AA Ni-Cd rechargeable batteries, or from a 9VDC, 300 mA adapter. The adapter can recharge the batteries and power the unit simultaneously. Should a different mains adapter from that supplied be used, please note that the **LM1** is indifferent to the polarity of the incoming DC. However, care must be taken to ensure that the mains adapter is rated at 9 VDC, with at least 300 mA capacity. Should the Ni-Cd batteries need replacing, remove the two screws on either side of the housing and carefully slide the assembly from the housing.

NOTE: USE Ni-Cd AA CELLS ONLY! THE USE OF OTHER TYPES OF BATTERIES COULD DAMAGE THE LM1 AND BE DANGEROUS.

The unit will operate for four hours continuously from a full charge, and require eighteen hours to recharge from a full discharge. It is recommended that the batteries are fully discharged before charging to improve battery life and efficiency.

SPECIFICATIONS

SPECIFICATIONS	DIGITAL	ANALOGUE
Input Connections	AES/EBU transformer balanced on 3 pin XLR connector	2 inputs (L&R) balanced on ¼" A gauge (stereo) jacks
Input Impedance	Selectable: "HiZ" or 110Ω	Greater than 20KΩ
Input Level	200 mVpp to max 7 Vpp	+18 dBu (before clip)
D/A Converter	18 bit resolution	
Sample Frequency Range	30 kHz to 50 kHz	
Frequency Response	20 Hz to 20 kHz ±0.5dB @ 48kHz sampling rate	20Hz to 20kHz ±0.5dB
Headphone Output	¼"stereo jack (600 ohm headphones)	

Audio Meters:
12 LED Stereo bargraph

Standard Factory Calibration:
Analogue Mode: -21 dBu to +18 dBu
Digital Mode: -39 dB FS to 0 dB FS

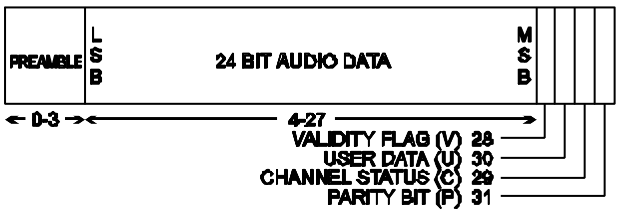
APPENDIX - DIGITAL AUDIO OVERVIEW

In order to interpret the information displayed when using the **LM1** to monitor digital audio signals, it may be beneficial to have some understanding of the AES/EBU digital audio standards and the transmission techniques employed.

The bit stream

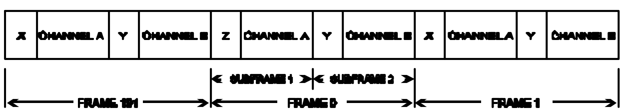
The recommended interface for serial transmission of linearly represented stereo audio data is defined by the AES/EBU (Audio Engineering Society/European Broadcast Union) standard, AES3-1985 and later amendments. The standard allows for audio data transmission of 16 to 24 bits sampled at a rate of 30 to 50 kHz. This data is organised as sub-frames, frames and channel status blocks.

FIG 1 SUBFRAME



As illustrated in fig.1, a sub-frame consists of a preamble, 24 audio data bits, validity bit (V), user data bit (U), channel status bit (C) and sub-frame parity bit (P). Each stereo sample or frame comprises two sub-frames, A and B. Sub-frames A and B represent the left and right programme information of the stereo sample respectively.
The validity bit indicates whether the audio sample is suitable for conversion to analogue. The **LM1** Validity Bit indicator warns you when this bit is set. The

FIG 2 FRAME FORMAT



LM1 also tests the sub-frame parity bit and will warn you if an error is detected.

A block structure is used to convey information in the (C) and (U) bits. A block consists of 192 frames, as shown in fig. 2. The 192 channel status (C) and user bits (U) are assembled at the end of the block to form 24 bytes of user and channel status information per channel. The channel status block carries information associated with the audio channel. Examples of information in the block are pre-emphasis, sampling rate, length of audio sample etc. The 24th byte of the channel status block carries the CRC (cyclic redundancy check) value that is used to test the validity of the entire channel status data block upon reception. The **LM1** will calculate the CRC of the first 23 bytes of the channel status block and compare it to the received value in byte 24. If the CRC does not match, the **LM1** indicates a CRC error.

Transmission

The serial data is coded in a bi-phase mark format (see fig 3). This format effectively changes the digital data from logic levels into transitions. A logic 1 level causes a transition within the bit period, whilst the level 0 will not. Transitions always separate each bit period. This bi-phase format has the advantage of being dc free and polarity insensitive. It should be noted that the preamble sync bits are unique in that they violate this coding format, i.e. they do not have transitions on all of the bit boundaries. If this coding structure is violated elsewhere, the **LM1** indicates a bi-phase error.

FIG 3 CHANNEL CODING

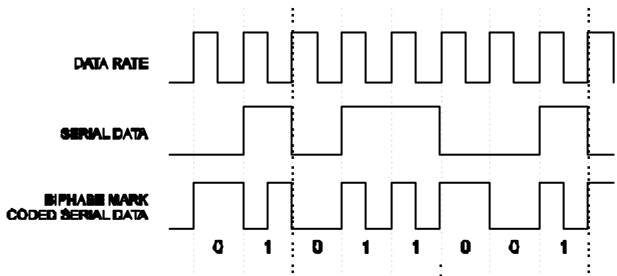
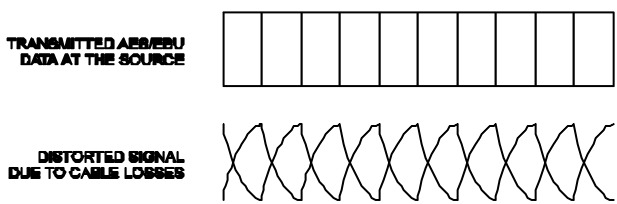


FIG 4 SIGNAL DISTORTION DUE TO CABLE LOSSES



From the description of the digital audio interface, it can be determined that every stereo sample requires 64 bits of data. At a sample rate of 48kHz, this translates into a bit rate of 3.072MHz.

For balanced AES/EBU transmissions, 110Ω shielded, twisted pair cable is recommended and for unbalanced transmissions, 75Ω co-axial cable. As rectangular pulse digital signals are inherently rich in harmonics, they require a wide bandwidth for distortion-free transmission.

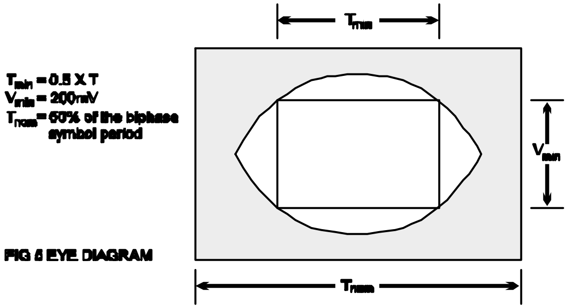


FIG 5 EYE DIAGRAM

The low pass filter effects of the cable causes the signal to disperse in time as cable lengths increase, (see fig 4). As shown in the eye diagram of fig.5, the AES/EBU standard dictates that a receiver shall correctly sense data with a minimum voltage of 200 mV and a width equal to 25% of the bi-phase symbol rate. An eye pattern can be seen on an oscilloscope by triggering it so that the pulses overlap. When a signal approaches these limits, the **LM1** indicates a confidence error as a warning that it may not be able to receive the signal correctly.