

# Warranty Information

**DISCLAIMER OF WARRANTIES:** Products manufactured by Linear Acoustic are warranted against defects in material and workmanship under the standard Telos Alliance 5-year warranty from the date of purchase. **THERE ARE NO OTHER IMPLIED OR EXPRESS WARRANTIES AND NO WARRANTY FOR MERCHANTABILITY OR FITNESS FOR A PARTICULAR PURPOSE.** 

During the warranty period Linear Acoustic Inc. will repair, or at our discretion replace, components which prove to be defective, provided the unit is returned, shipped pre-paid to us directly with a return authorization (RA) number clearly marked on the packaging. **Please note, this RA number must be present or package will be refused and returned to sender.** 

All requests for repairs MUST include the unit serial number to ensure quick and accurate service.

#### DEFECTS CAUSED BY UNAUTHORIZED MODIFICATIONS, MISUSE OR ACCIDENTS, UNAUTHORIZED CUSTOMER REPAIRS, OR ANY FURTHER DAMAGE CAUSED BY INADEQUATE PACKAGING FOR SERVICE RETURN ARE NOT COVERED BY THIS WARRANTY.

#### PLEASE SAVE THE SHIPPING CARTON AND ALL PACKING MATERIALS. FAILURE TO RETURN UNIT IN ORIGINAL SHIPPING CARTON AND PACKING MATERIALS WILL RESULT IN A CHARGE FOR NEW SHIPPING MATERIALS.

**LIMITATION OF PERIOD OF ACTION ON CONTRACT:** No action, regardless of form, arising out of the transactions under this agreement may be brought by buyer, its successors, agents and/or assigns, more than three years from date of purchase.

**LIMITATION OF LIABILITY:** It is understood and agreed that the liability of Linear Acoustic, whether in contract, in tort, under any warranty, in negligence or otherwise shall not exceed the cost of repair or replacement of the defective components and under no circumstances shall Linear Acoustic be liable for incidental, special, direct, indirect or consequential damages, or loss of use, revenue or profit even if Linear Acoustic or its agents have been advised, orally or in writing, of the possibility of such damages.

This product contains Dolby Digital (AC-3), Dolby Digital Plus and Dolby E decoding and is manufactured under license from Dolby Laboratories, Inc.

Linear Acoustic, the "LA" symbol, UPMAX, AEROMAX, LQ-1, AERO.1000, and AERO.2000 are trademarks or registered trademarks of Linear Acoustic Inc., all other trademarks remain the property of their respective owners.

# **Regulatory Notices and Fusing Information**

### FCC

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

## Canada

This Class A digital apparatus complies with Canadian ICES-003.

### UL



WARNING: Troubleshooting must be performed by a trained technician. Do not attempt to service this equipment unless you are qualified to do so.

Check that the correct fuses have been installed. To reduce the risk of fire, replace only with fuses of the same type and rating.

Exposed portions of the power supply assembly are electrically "hot". In order to reduce the risk of electrical shock, the power cord MUST be disconnected when the power supply assembly is removed.

The ground terminal of the power plug is connected directly to the chassis of the unit. For continued protection against electric shock, a correctly wired and grounded (earthed) threepin power outlet must be used. Do not use a ground-lifting adapter and never cut the ground pin on the three-prong plug.



## UK

As the colours of the cores in the mains lead may not correspond with the coloured markings identifying the terminals in your plug, proceed as follows:

- The core that is coloured green and yellow must be connected to the terminal in the plug identified by the letter E or by the earth symbol <u></u> or coloured green or green and yellow.
- The core that is coloured blue must be connected to the terminal that is marked with the letter N or coloured black.

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# **Chapter 1: Introduction**

The Linear Acoustic LQ-1 provides comprehensive standardized loudness measurement with all-inclusive I/O. Standard SDI, AES, Optical, Analogue and DVB-ASI inputs are followed by decoding for PCM, Dolby or MPEG audio. The decoded audio is then measured by the Dolby loudness engine which supports BS.1770-1 and BS.1770-2 with Dialogue Intelligence automatic gating. The result is a metered value that agrees with standards around the world.

The LQ-1 includes the following features:

- ITU-R BS.1770-1/2/3 loudness metering
- Selectable Dolby<sup>®</sup> Dialog Intelligence<sup>™</sup>
- Decoding of Dolby Digital (AC-3), Dolby Digital Plus and Dolby E
- Full-time LtRt or LoRo downmix of main 5.1 channel program
- HD/SD-SDI Input, re-clocked output for embedded audio and VANC metadata
- Discrete AES Inputs
- Balanced stereo analog inputs and outputs
- LTC and ATC (SDI) timecode inputs for event time stamping
- Auto-ranging power supply, DC Input for available backup PSU

### Software Updates

The RJ-45 Ethernet connector is for software updates and SNMP remote monitoring.

### Warranty and Feedback

Please take a moment to fill out and return the included warranty card. This enables us to keep you informed when new software or documentation is available.

We are very interested in your feedback. LQ-1 was designed based on the creative and generous input gathered from many broadcast engineers, and it will continue to evolve thanks to ongoing suggestions and comments from our users. We look forward to hearing from you.

**Phone:** 717-735-3611 **FAX:** 717-735-3612 **Web:** www.LinearAcoustic.com This page is blank (or nearly so)

# Chapter 2: Installation and Setup

This chapter covers:

- Unpacking and inspecting for shipping damage
- Rear panel connectors
- Installation
- Getting started quickly

# 2.1 Unpacking and Inspection

Before unpacking the unit, inspect the outer carton for shipping damage. If the carton shows damage, inspect the unit in those areas. If it is not damaged, please save the carefully designed shipping carton and packing materials. Alternate packing materials may be inadequate and lead to damage not covered by the warranty.

The following items are provided in a bag:

- One IEC power cord (matches country of order)
- This user guide
- Small piece of SWAG
- Option: If purchased, a +12VDC external power supply with an IEC power cord

Please fill out and return the warranty card to Linear Acoustic so we can inform you when new software and documentation are available.

# 2.2 Rear Panel

All electrical I/O connections occur on LQ-1's rear panel. See Chapter 7: Specifications for specific connector pinouts. All rear panel connections are described below.



Figure 2-1 LQ-1 rear panel

### Metadata

RS485 metadata input per SMPTE RDD-6. Currently not supported.

### GPI/O

Up to 4 TTL-level inputs where Ground=Active. GPI enables recall of presets or reset of loudness meter integration. GPO enables indication of loudness over/under threshold, center channel level under threshold, silence detection, channels over threshold, reference loss, and bitstream errors.

### Ethernet

Automatically senses 10/100 BASE-T connections for SNMP monitoring and software updates.

### SDI In/Re-clock Out

Any of the 16 audio channels from an HD/SD-SDI signal can be de-embedded and used with, or instead of, the AES inputs. Channels can be selected in pairs. Also used for DVB-ASI input/re-clocked output.

### **Optical In**

Connect a 48 kHz AES plant reference signal to this connector (not video or word clock). The signal may be an AES Black or AES signal with audio. All Main Audio Inputs must be synchronous and locked to the AES reference signal applied here.

#### AES In

Connect 48 kHz PCM or Dolby-encoded AES signals to these inputs. The input channels are configured as follows:

1/2 = Left front/Right front
3/4 = Center/LFE
5/6 = Left surround/Right surround
7/8 = Stereo +2 or Local input (Not Currently Used)

### LTC In

LTC Timecode Input

### **AES Out**

A full-time two-channel downmix. Available selections include LtRt (matrix surround compatible) and LoRo (better stereo compatibility with no phase shifts applied to the surround channels).

### Analog In/Out

Balanced analog audio, +4dBu = -20dBFS. These

### +12VDC In

Redundant DC power input, +12VDC +/- 0.2 Volts, center positive. Connect optional external power supply here if purchased. Connector is threaded for locking.

### PSU

Apply 100–240 VAC filtered power. Can be used in tandem with the +12VDC backup power supply. The two inputs are *summed*, thus improving plant reliability and lowering the heat generated. Only one supply is required for continuous operation, but the supplies operate in parallel to share the load.

**WARNING:** We STRONGLY recommend using an uninterruptable AC power supply (UPS) either locally or as part of the facility.

## 2.3 Installation

LQ-1 installation requires:

- One standard rack space with adequate ventilation
- Standard 75-Ohm BNC cables

To connect digital equipment with 110-Ohm XLR connectors, use impedance-matching transformers (available from Canare and Neutrik).

• Proper selection of clock reference signal

The unit defaults to internal 48 kHz, but should be set to AES Ref In, AES 1, or SDI depending on how the unit will be used. Note that if SDI embedding is enabled, Reference will automatically be forced to SDI.

# 2.4 Getting Started Quickly

LQ-1 is configured at the factory, and is ready to go on the air after this brief setup procedure.

### Powering Up and Shutting Down

- 1. Connect AC power to the power supply, the unit will automatically power on.
- 2. Remove the AC (and backup DC if present) power connection and the unit will, unsurprisingly, power down.

### Quick Setup

- **1.** Connect AC power and power on LQ-1 according to the section above.
- 2. Apply audio sources to Main AES inputs, the SDI Input, or the Optical Input.
- 3. Select the appropriate reference via the front panel user interface
- 4. Select the appropriate input signal via the front panel user interface

You should now have audio on the Main AES output, the analog output and the front panel head-phone connection.

# **Chapter 3: Applications and Metering**

# 3.1 Applications

The following pages show how to integrate LQ-1 within several different signal environments. The setup steps are the same for each.

### AES

When used in AES mode, discrete digital audio is applied to the unit for metering. It is also possible to connect an encoded signal to the first AES input. Note that the AES input is normally also compatible with the IEC 61937 digital output of consumer gear, although a better option would be to use the TOSLINK optical input for these situations.

- Select Inputs: Proc In 1/2 = AES 1/2; In 3/4 = AES 3/4; In 5/6 = AES 5/6
- Select Clock Source: AES Input 1
- Select Meter mode (i.e. set up the display): user preference
- Select logging mode: none or user preference



Figure 3-1 Typical AES application.

### **TOSLINK (Optical)**

Useful for connecting to consumer set top boxes or other sources of digital audio. This is the ideal way to measure signals from cable, satellite, and even the Internet. Many computers have a TOSLINK optical output, and most every modern Apple computer will have an optical output as part of the head-phone connector. A widely available 1/8" optical to TOSLINK cable would be required.

- **NOTE:** When connecting to the digital audio output of a computer, make sure that the computers volume is set to Unity. On Windows machines, this can be found by right clicking on the speaker near the system clock, then selecting "Audio Playback Devices" and on an Apple, while this is normally set automatically when the connector is detected, settings can be found under Audio/MIDI setup.
  - Select Inputs: Proc In 1/2 = TOSLINK
  - Select Clock Source: TOSLINK
  - Select Meter mode (i.e. set up the display): User preference
  - Select logging mode



Figure 3-2 Typical TOSLINK optical application.

### SDI

Audio can be extracted from an applied HD or SD signal.

- Select Inputs: Proc In 1/2, 3/4, and 5/6 to the appropriate SDI pairs
- Select Clock Source: SDI
- Select Meter mode (i.e. set up the display): user preference
- Select logging mode: None, or user preference



Figure 3-3 Typical SDI application.

### **DVB-ASI**

Dolby Digital (AC-3) can be extracted directly from a DVB-ASI signal, the audio can then be decoded and measured. This is useful to monitor directly from a local encoder or via an off-air receiver to check the final transport stream.

- Select Inputs Proc In 1/2 = SDI 1/2
- Select Clock Source SDI
- Select DVB-ASI Mode Enable
- Select audio PID to extract Hex, 0x0034 is a good start
- Select Meter mode (i.e. set up the display)
- Select logging mode



Figure 3-4 DVB-ASI application.

# 3.2 Loudness Metering

The LQ-1 Loudness Meter employs a standardized measurement method per ITU-R BS.1770 and can be coupled with Dolby Dialogue Intelligence<sup>TM</sup> technology to allow the LQ-1 to automatically measure only the dialogue portions programming. These methods are used to accurately determine the dialogue normalization metadata value for Dolby Digital, Dolby Digital Plus, and Dolby E broadcast content.

Both ITU-R BS.1770-1 and BS.1770-2 algorithms estimate loudness by computing the frequency weighted (k) energy average over time similar to Leq(A), and each algorithm produces a single value representing the overall loudness level. The frequency weighting for both BS.1770-1 and BS.1770-2 is based on two filters in cascade: a pre-filter (a high-frequency shelving filter), followed by Leq(RLB), a revised low B-weighting filter (essentially a high-pass filter). The ITU-R BS.1770-2 loudness measurements uses a loudness gating method to better estimate the perceived loudness of the signal. The gating function is driven by a 400 ms moving average, updated every 100 ms to provide a 75% overlap between successive gating blocks. The loudness is then estimated using a -70 dBFS absolute (safety) gate and a -10 dB relative gate threshold. The short-term loudness measurement is ungated.

## 3.2.1 Dolby Dialogue Intelligence

Dialogue Intelligence allows the LQ-1 to automatically base ITU-R BS1770-1 measurements on the portions of the input signal that contain the characteristics of dialogue. This feature provides users at all skill levels with the capability to easily quantify the level of dialogue within broadcast programs.

Dialogue Intelligence is very useful for estimating the loudness of long-form programs, especially programs with wide dynamic range. These programs, when measured with infinite integration, can result in misleading loudness values. Dialogue Intelligence focuses the metering on the dialog in a program, if present, regardless of the dynamics of surrounding content. This results in a value that more closely matches human perception of the average loudness.

A common misconception about Dialogue Intelligence is that if no dialog is present then no loudness value is produced. The LQ-1 enables loudness metering to continue by falling back to the non-dialog-gated BS.1770-1 or BS.1770-2 measurement which is always running in parallel.

Figure 2-3 below shows how Dialogue Intelligence is combined with ITU-R BS.1770 metering.



Figure 3-5 Implementation of Dialogue Intelligence with BS.1770-1/2 metering. Note that dialog (speech) gated and ungated measurements run in parallel and are always active. (Drawing courtesy of Dolby Laboratories)

### 3.2.2 Measurement Methods

The LQ-1 has two different methods of operation for the measurement period: Infinite and Short Term.

Short Term	Infinite
Live broadcast event, provides dynamic feedback to the audio mix engineer	Program Ingest
Postproduction, provides dynamic feedback to the audio mix engineer	Postproduction, to ensure entire program meets delivery specifications
Quality control: Measuring short term program dynam- ics (or the short term dynamics in dialogue level when Dialogue Intelligence is enabled)	Quality control: Measuring the overall program for conformance with delivery specifications
Logging short term loudness history to spot trends or non-compliant segments	Logging long term loudness history to spot trends or non-compliant segments; vastly more useful with Dialogue Intelligence enabled
Calibration of analog cable head ends, specifically modulator deviation adjustments	

### Short Term Measurement

The Short Term measurement method displays a measurement value for the previous three or ten seconds as a sliding window. When set to ten seconds, the first measurement value displayed corresponds to 0–10 seconds, the next to 1–11 seconds, the next to 2–12 seconds, and so forth. When set to three

seconds, the first measurement value displayed corresponds to 0-3 seconds, the next to 1-4 seconds, the next to 2-5 seconds and so forth.

As the short term measurement only considers the last three or ten seconds of program material, the measurement value has the potential to be much more dynamic than if measured with the infinite method. Highly processed channels and programming will most likely not exhibit this behavior. The benefit of this measurement method is that it allows the operator to see short term variations within a program in loudness level or in dialogue level when Dialogue Intelligence is enabled. Many skilled audio operators prefer to use the short term measurement, as they find the dynamic information to be very useful when mixing or producing a program. The short term method is also very useful for measuring and logging the loudness history of a given program during the QC, post production process, or a particular television service or channel in a cable head-end facility.

#### **Infinite Measurement**

Infinite measurements cover the entire period since the measurement was last reset. The ITU-R BS.1770-1, ITU-R BS.1770-2, and Leq(A) algorithms are all available using the infinite method. Enabling Dialogue Intelligence allows you to simply quantify all the sections of the program that contain only dialogue and use the ending measurement value for analysis and normalization. This value is also commonly used to set the dialogue normalization parameter within Dolby Digital and Dolby E bit-streams.

The infinite method is typically used when it is possible to measure the entire duration of the program (for example, all 30 seconds of a commercial or the whole two hours of a movie). It provides the most accurate measurement. This method is most often used in ingest, QC, and post production applications where audio metadata is being authored and levels can, in most cases, be controlled and adjusted.

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# Chapter 4: Menus

This chapter discusses how to use the front panel OLED interface to access the menus. Most of it is rather obvious, and a bit of experimentation will quickly make you comfortable navigating through the sub-menus. The deeper sub-menus in the hierarchy control more detailed and complex functionality.

The controller has 5 main states which can be activated by "rocking" the knob up, down, left, or right, or by pushing it in.

- · Right Same as Next or Enter
- · Left Same as Back or Escape
- · Up, Down As you would expect
- Rotary Parameter adjustment

Push to reset loudness meter integration

When a parameter is changed, an asterisk (\*) will be displayed next to the parameter denoting that it has changed and that in order for this change to be stored, the right (Next) front panel button must be pressed. To continue without storing the change, press the left (Prev) front panel button.

Like all of our products, when in doubt you can simply use the Left Arrow (Prev) to back out towards the Main menu level.

## 4.1 Main Menu Level

Referring to the menu tree on the preceding page, it can be seen that the Main Menu level can be scrolled through using either the left or right arrows. The choices are: Version, Status, Statistics, and Setup, each of which have sub menus (except the version screen). The sub menus can be accessed by pressing the down arrow. Note that both the Status and Statistics screens are informational and have no adjustments.

# 4.2 Detailed Menu Functions and Setup

Below are in-depth descriptions of each of the menu settings.

### **Menu Settings**

As with other Linear Acoustic products, the menus are structured so that top level menus provide current system status information, while setup is performed by accessing lower level menus. In other words, the deeper you navigate, the more detailed the status and setup information becomes. In general, the most used information and setup parameters are kept towards the top level.

### **Top Level Menus**

The default screen upon boot up is the Main screen. Additional Top menus are located around the Main screen and can be accessed with the Left and Right arrows:

<-> SDI Status <-> Main <-> Data <-> Statistics <-> Setup <-> Version <->

Each of these top level menu screens will be described below.

**SDI Status -** Details of applied SDI signal including type and frame rate.

**Main Screen -** Default top level screen displays signal levels, adjustable primary and secondary loudness or true peak metering, codec in use, key metadata information like dialnorm and audio coding mode (acmod), speech detection. It also has the standard Linear Acoustic "floating bars" which offer a quick indication of loudness versus dialnorm.



Below is the main screen showing a Dolby Digital program:



**Data -** Displays loudness data and statistics, as well as audio metadata if present (i.e. if the input signal is Dolby Digital, Dolby Digital Plus, or Dolby E).

**Statistics -** Shows current installed firmware, options, and receive CRC errors for AES and metadata signals.

**Setup -** Down to access sub-menus for configuring the operation of the unit. See 4.2.1 below for specific details.

Version - Displays the current software version of the LQ-1.

### 4.2.1 LQ-1 Setup: Presets

- User Preset Up to sixteen presets can be stored and recalled and this menu allows selection of the current preset for editing. The first six are factory presets:
  - **AES\_1\_8** PCM audio from the four AES inputs is used. Note that SRC Bypass is disabled so encoded audio will not be decoded.
  - **SDI\_1\_8 -** PCM audio will be extracted from the first four pairs of an applied SDI signal and used for metering. Note that SRC Bypass is disabled so encoded audio will not be decoded.
  - **SDI\_9\_16** PCM audio will be extracted from the last four pairs of an applied SDI signal and used for metering. Note that SRC Bypass is disabled so encoded audio will not be decoded.
  - **AES\_1\_8\_ENC -** PCM or encoded audio from the four AES inputs is used, encoded audio will be decoded since SRC Bypass is enabled.
  - **SDI\_1\_8\_ENC** PCM or encoded audio will be extracted from the first four pairs of an applied SDI signal, decoded if necessary and used for metering.
  - **SDI\_9\_16\_ENC -** PCM or encoded audio will be extracted from the last four pairs of an applied SDI signal, decoded if necessary and used for metering.
- **Proc In 1/2 through 7/8** Selects the signal pairs that map to the 8 inputs of the metering core pair-by-pair.
  - AES In 1/2-7/8 Audio from any of the four BNC inputs is used
  - TOSLINK In Audio from the Optical input is used
  - Analog XLR In Audio from the stereo analog inputs is used
  - SDI 1/2-15/16 Audio channel pairs from applied HD-SDI signal
- Clock Source Select AES In 1, SDI, TOSLINK or Internal 48kHz.
- **DVB-ASI** Enables DVB-ASI mode when enabled. Signal is applied to the SDI input, specified PID is extracted, Dolby Digital (AC-3) decoded and metered. A re-clocked DVB-ASI signal is output from SDI Out.
- **DVB-ASI PID** Specify (in hex) the audio PID to be extracted.
- **SRC Bypass** If the input signal and clock source are locked, audio sample rate conversion for the SDI and AES output is not required and can be bypassed to lower throughput latency by about 3msec. Default is Enabled (i.e. SRC is bypassed), but as it is part of the preset structure, it can change.

NOTE: SRC Bypass MUST be enabled (i.e. SRC is bypassed) for encoded audio to be decoded. For DVB-ASI and TOSLINK, SRC Bypass should always be disabled.

- Apply DD/DD+ DRC Line Mode or RF Mode DRC can be selected to be applied to decoded Dolby Digital or Dolby Digital Plus audio. Note that dialnorm will also be applied, resulting in audio that is normalized to -31dBFS in Line Mode and -20dBFS in RF Mode.
- Apply DE DRC Line Mode or RF Mode DRC can be selected to be applied to decoded Dolby E audio. Note that dialnorm will also be applied, resulting in audio that is normalized to -31dBFS in Line Mode and -20dBFS in RF Mode.

### 4.2.2 LQ-1 Setup: I/O

This menu allows for general configuration of input and output signals. These settings are normally adjusted during installation and will normally not need to be changed.

- **Master Bypass -** Bypasses all digital audio, SDI, and metadata signals from input to output via relays. Same as powering the unit off.
- **Output Source -** Selects the source for the Main AES output BNC. Choices are:
  - Downmix
  - TOSLINK In
  - Analog XLR In
  - SDI In 1/2-15/16
- **Downmix Type -** Selects LtRt or LoRo downmix of applied program. Default is LoRo.
- Headphone Boost Enabling adds +6dB of gain to the headphone output; useful for monitoring quiet signals reduced in level because of dialnorm. Default is Enabled.
- **Clock Src Type -** If the clock source for the unit is from a stable reference, choose stable. For instances where clock reference might switch, choose Changing to minimize (but not eliminate) audio corruption. We strongly recommend using a stable reference source!
- Validity Bit Sets the channel status bits on the AES output to indicate whether a signal is Valid for analog to digital conversion or is encoded data that should not be decoded. Default is Valid.

- Audio Bit Sets the channel status bits on the AES output to indicate whether a signal is Audio Data (PCM) or Non-Audio Data (encoded). Default is Audio Data
- **GPI Control** Enables GPI control of unit allowing for remote recall of up to four presets or reset of loudness meter integration.
- **GPI 1-4 Function -** Configures each of the four general purpose inputs. Currently, the GPIs allow for recall of any stored preset.
- **GPI 1-4 Preset -** Allows selection of the preset that will be recalled by the GPI.
- **GPO 1-4 Alert When-** Configures each of the four general purpose outputs:
  - Never GPO does nothing.
  - Follow GPI Each GPO will match the state of the corresponding GPI, so if GPI 1 is commanded Low, GPO1 will follow.
  - LKFS Over Thr GPO will be active if the currently monitored input audio exceeds the adjustable threshold described below.
  - LKFS Under Threshold GPO will be active if the currently monitored input audio is below the adjustable threshold described below.
  - Center Ch Under GPO will be active if the Center channel of the input audio is not present, i.e. loss of Center.
  - Silence Detect GPO will be active if the all channels of the input audio are not present, i.e. loss of audio.
  - Auto Ch Under GPO will be active if the Center channel of the input audio is not present when program is indicated as being 5.1 via metadata, else GPO will be active if all channels are not present.
  - Any Chan Over GPO will be active if any applied and monitored audio channel clips.
  - Reference Loss GPO is active if reference signal is missing. If the unit is set to use PCM via BNC or Dolby via BNC, then Main In 1/2 is reference, and if the unit is set for PCM or Dolby via SDI then the SDI input is reference.
  - Bitstream Error GPO is active if a CRC error is detected on an applied audio bitstream (Dolby Digital/Plus or Dolby E).
- **GPO 1-4 Threshold -** Adjustable from Mute to 0dB, default is -40dB.

- **GPO 1-4 On Delay -** Adjusts how long to wait before triggering the GPO. Default is 0 seconds, adjustable to 20 seconds. As an example, if set to 0 seconds, any event that would trigger GPO will do so instantly. If set to 20 seconds, the event must occur for 20 seconds for the GPO to change. This is useful when indicating loss of Center channel, where pauses are OK but loss for 20 seconds may not be.
- **GPO 1-4 Off Delay -** Adjusts how long to hold a GPO function once triggered. Think of it like a pulse stretcher. Allows for longer indications of quick events such as loss of metadata.

## 4.2.3 LQ-1 Setup: Meter Setup

Configure the metering algorithm, integration time, and enable Dialogue Intelligence if desired.

- Loudness Mode Select meter type. Six choices are:
  - Leq(A)/Leq(A)+D The original A-weighted loudness measure, included for completeness. Very close in performance to BS.1770-1 which is actually Leq(RLB). Select +D to enable Dialogue Intelligence for speech gating of the loudness measurement.
  - BS.1770-2/1770-2+D -
  - BS.1770-1/1770-1+D -
- **10s Speech Gate** Select between Continuous and Hold on Non-Speech. When speech is not detected, as indicated by the reverse video "S" on the display turning off, this selection will determine what the meter does. For Continuous, the non-speech gated 10-second value for the loudness mode selected will replace the speech gated value. For Hold, the last valid speech gated value will be frozen in the display. Default is Continuous.
- Main Loudness Type Select which integration time will be displayed on the Main loudness section of the main screen (to the right of the dialnorm value). Default is 10sec Loudness.
- Sec Loudness Type Select which integration time will be displayed below the loudness and dialnorm readings on the main screen. Default is Max True Peak.
- **Graph Loud Type -** Select which integration time will drive the floating bar indication on the main screen (on top of the level meters). Default is 3sec Loudness.
- Log Loud Type Select which integration time will be the source for logging data. Default is Integrated Loudness.

- Log Frequency Select how often to set an SNMP trap with a loudness value. Default is No Logging.
- Log Timecode Src Select between LTC and ATC timecode to accompany logging data. Default is LTC.

### 4.2.4 Communication

Set unit main IP address, Subnet, and dual SNMP Trap addresses.

### 4.2.5 System

Allows TCP/IP settings to be restored to factory default.System

	System Down To View	Set to Defaults Set: IP Comm																			
I Menu Tree	Communication Down To View	IP Address 192.168.0.20	Net Mask 255.255.255.0	Port Number 5700	MAC Address 0050:C2:51:25:EE	SNMP Trap Addr 1 0.0.0.0	SNMP Trap Addr 2 0.0.0.0														
Acoustic LQ-	Meter Setup Down To View	Loudness Mode BS1770-2+D	10s Speech Gate Continuous	Main Loud Type 10Sec Loudness	Sec Loudness Type Max True Peak	Graph Loud Type 3Sec Loudness	Log Loud Type Integ Loudness	Log Frequency No logging	Log Timecode Src LTC												
Linear	Down To View	Master Bypass Disable	Output Source TOSLINK In	Downmix Type LoRo	Headphone Boost On	Clock Src Type Stable	Validity Bit Valid	Audio Bit Audio data													
tulinam factor			Preset Name Down To View	Edit Preset Name AES_1_8																	
10-1 Setter Boson To Visa	+ Presets Down To View ►	User Preset AES_1_8	Preset Params Down To View	Proc In1/2 Src AES In 1/2	Proc In 3/4 Src AES In 3/4	Proc In 5/6 Src AES In 5/6	Proc In 7/8 Src AES in 7/8	Clock Source AES Input 1	DVB-ASI Disable	DVB-ASI PID 0x0034	SRC Bypass Enable	Apply DD/+ DRC Line Mode + DN	Apply DE DRC Bypass	GPI Control Disable	GPI 1(to 4) Function None	GPI 1 (to 4) Preset AES_1_8	GPO 1 (to 4) Alert When Never	GPO 1 (to 4) Threshold -40 dB	GPO 1 (to 4) On Delay 0 Sec	GPO 1 (to 4) Off Delay 0 Sec	
ини) 10.1.1 Знатестот Down To Vian	Firmware Version 5700 00 11	Device Options SDI In On	App Uptime DD:HH:MM:SS	GPI Function (Disabled) 1 None 2 None	3 None 4 None A None There Make	GPC No Arent Insen value 1 0 Never 1 0 Never 1 0 Never	Ref Chg Num=xxx	LTC:: Presence fr atc	To Reset LKFS: Hit TAKE	SNMP Trap Errs Num=xx Code=xxxx	SNMP Tx Errs Num=xx Code=xxxx	SNMP Rx Errs Num=xx Code=xxxx	SNMP Pkt Rx=0000 Trp=0000 Tx=0000	CN1100 Status Detect=Y Rdy=Y	CN1100 FW Vers. V1.0.0.2	CN1100 HW Vers. V0.0.0.3	CN1100 Serial No. XXX	CN1100 CRC Errs DE=000 DD/+=000	CN1100 Tx Errs Num=00 Code=0000	CN1100 Rx Errs Num=00 Code=0000	AES In 12345678 Detect •
Main Screen																					
0 Nutati																					

# Chapter 5: SNMP and Logging

# 5.1 SNMP Setup

The LQ-1 supports error reporting via the Simple Network Management Protocol. Status of critical functions such as the health of each power supply, codec (Dolby decoder) function, audio presence and plus the status of each of the four user-configurable GPO outputs are reported.

The current LQ-1 MIB file is copied below. Note that this data can be copied and pasted to a text file that can be re-named LINEAR-ACOUSTIC-LA5700.mib and used by an SNMP manager.

LINEAR-ACOUSTIC-LA5700-MIB DEFINITIONS ::= BEGIN -- MIB for Linear Acoustic la5700 devices IMPORTS enterprises FROM RFC1155-SMI FROM RFC-1215 TRAP-TYPE OBJECT-TYPE FROM RFC-1212; linearAcoustic OBJECT IDENTIFIER ::= { enterprises 28660 } la5700 OBJECT IDENTIFIER ::= { linearAcoustic 5700 } la5700-system OBJECT IDENTIFIER ::= { la5700 1 } la5700-status OBJECT IDENTIFIER ::= { la5700 2 } -- la5700-system \_\_\_ -- model -model OBJECT-TYPE SYNTAX OCTET STRING (SIZE (0..40)) OBJECT-TYPE la-model ACCESS read-only STATUS mandatory DESCRIPTION "Linear Acoustic Model." ::= { la5700-system 1 } -- software version software-version OBJECT-TYPE SYNTAXOCTET STRING (SIZE (0..40))ACCESSread-onlySTATUSoptional DESCRIPTION "Device's software version." ::= { la5700-system 2 } -- firmware version firmware-version OBJECT-TYPE 
 SYNTAX
 OCTET STRING (SIZE (0..40))

 ACCESS
 read-only

 STATUS
 optional
 DESCRIPTION "Devices's firmware version." ::= { la5700-system 3 } -- la5700-status

```
-- Power Supply 1
power-supply-1 OBJECT-TYPE
   SYNTAX INTEGER {fail (1), ok (2)}
              read-only
mandatory
   ACCESS
   STATUS
   DESCRIPTION
       "Status of power supply 1."
   DEFVAL { 1 }
    ::= { la5700-status 1 }
-- Power Supply 2
power-supply-2 OBJECT-TYPE
             INTEGER {fail (1), ok (2)}
   SYNTAX
    ACCESS
              read-only
   STATUS
              optional
   DESCRIPTION
        "Status of power supply 2."
   DEFVAL { 1 }
    ::= { la5700-status 2 }
-- Master Bypass
master-bypass OBJECT-TYPE
   SYNTAXINTEGER {off (1), on (2)}ACCESSread-only
   STATUS
              optional
   DESCRIPTION
       "Status of Master Bypass."
   DEFVAL { 1 }
   ::= { la5700-status 3 }
-- System Reference
system-reference OBJECT-TYPE
   SYNTAX INTEGER {absent (1), present (2)}
   ACCESS
              read-only
           optional
    STATUS
   DESCRIPTION
       "Status of System Reference."
   DEFVAL { 1 }
   ::= { la5700-status 4 }
-- Audio Detected (Primary)
pri-audio-detected OBJECT-TYPE
    SYNTAX
              INTEGER \{no (1), yes (2)\}
   ACCESS
              read-only
    STATUS
              optional
    DESCRIPTION
       "Detection of Primary Audio Signal."
   DEFVAL \{1\}
    ::= { la5700-status 5 }
-- GPO1 Current State
gpol-status OBJECT-TYPE
   SYNTAX
            INTEGER \{off (1), on (2)\}
   ACCESS
             read-only
   STATUS
              optional
   DESCRIPTION
       "Status of GPO1."
   DEFVAL \{1\}
    ::= { la5700-status 6 }
-- GPO2 Current State
gpo2-status OBJECT-TYPE
            INTEGER {off (1), on (2)}
   SYNTAX
   ACCESS
              read-only
              optional
    STATUS
   DESCRIPTION
       "Status of GPO2."
   DEFVAL { 1 }
    ::= { la5700-status 7 }
-- GPO3 Current State
```

```
gpo3-status OBJECT-TYPE
   SYNTAX
            INTEGER \{ off (1), on (2) \}
   ACCESS
               read-only
   STATUS
              optional
   DESCRIPTION
       "Status of GPO3."
   DEFVAL { 1 }
   ::= { la5700-status 8 }
-- GPO4 Current State
gpo4-status OBJECT-TYPE
   SYNTAX
               INTEGER {off (1), on (2)}
   ACCESS
               read-only
   STATUS
               optional
   DESCRIPTION
       "Status of GPO4."
   DEFVAL { 1 }
   ::= { la5700-status 9 }
-- GPO1 On Current Assignment
gpol-on-assignment OBJECT-TYPE
              OCTET STRING (SIZE (0..40))
   SYNTAX
               read-only
   ACCESS
   STATUS
              optional
   DESCRIPTION
       "Assignment of GPO1 when ON."
   ::= { la5700-status 10 }
-- GPO1 Off Current Assignment
gpol-off-assignment OBJECT-TYPE
   SYNTAX
               OCTET STRING (SIZE (0..40))
               read-only
   ACCESS
   STATUS
               optional
   DESCRIPTION
       "Assignment of GPO1 when OFF."
   ::= { la5700-status 11 }
-- GPO2 On Current Assignment
gpo2-on-assignment OBJECT-TYPE
   SYNTAX
            OCTET STRING (SIZE (0..40))
   ACCESS
               read-only
               optional
   STATUS
   DESCRIPTION
        "Assignment of GPO2 when ON."
   ::= { la5700-status 12 }
-- GPO2 Off Current Assignment
gpo2-off-assignment OBJECT-TYPE
   SYNTAX
               OCTET STRING (SIZE (0..40))
   ACCESS
               read-only
   STATUS
               optional
   DESCRIPTION
       "Assignment of GPO2 when OFF."
   ::= { la5700-status 13 }
-- GPO3 On Current Assignment
gpo3-on-assignment OBJECT-TYPE
   SYNTAX
              OCTET STRING (SIZE (0..40))
   ACCESS
               read-only
   STATUS
               optional
   DESCRIPTION
        "Assignment of GPO3 when ON."
   ::= { la5700-status 14 }
-- GPO3 Off Current Assignment
gpo3-off-assignment OBJECT-TYPE
               OCTET STRING (SIZE (0..40))
   SYNTAX
   ACCESS
               read-only
   STATUS
              optional
   DESCRIPTION
       "Assignment of GPO3 when OFF."
   ::= { la5700-status 15 }
```

```
-- GPO4 On Current Assignment
gpo4-on-assignment OBJECT-TYPE
   SYNTAX
              OCTET STRING (SIZE (0..40))
   ACCESS
               read-only
   STATUS
               optional
   DESCRIPTION
      "Assignment of GPO4 when ON."
   ::= { la5700-status 16 }
-- GPO4 Off Current Assignment
gpo4-off-assignment OBJECT-TYPE
   SYNTAX
              OCTET STRING (SIZE (0..40))
   ACCESS
               read-only
   STATUS
               optional
   DESCRIPTION
       "Assignment of GPO4 when OFF."
   ::= { la5700-status 17 }
-- Encoder/Decoder
codec
             OBJECT-TYPE
               INTEGER {error (1), ok (2)}
   SYNTAX
   ACCESS
             read-only
   STATUS
              optional
   DESCRIPTION
       "Status of Encoder/Decoder module."
   DEFVAL { 1 }
   ::= { la5700-status 18 }
-- Loudness Log record
loudness-log-rec OBJECT-TYPE
              OCTET STRING (SIZE (0..255))
   SYNTAX
   ACCESS
              read-only
   STATUS
              optional
   DESCRIPTION
        "Loudness Log record."
   ::= { la5700-status 19 }
-- Loudness Log Frequency
loudness-log-freq OBJECT-TYPE
   SYNTAX
              INTEGER (0..60)
   ACCESS
              read-only
   STATUS
              optional
   DESCRIPTION
       "Loudness Log Frequency in seconds. Zero indicates No Logging."
   DEFVAL { 0 }
   ::= { la5700-status 20 }
_ _
-- traps
-- 12/07/2012 RDC
-- Linear Acoustic rules for enterprise-specific traps
-- 1. Use trap numbers from 10 to 127.
     This makes them different from generic traps (0 - 5) and
___
--
     lets them fit into a consistent and small space in BER TLV packet.
-- 2. Use VARIABLES sparingly to keep the overall BER packet
___
     size small and processing down.
-- Power Supply 1 failed
trap-psl-failed TRAP-TYPE
   ENTERPRISE la5700
   DESCRIPTION
        "Trap, Power Supply 1 failed."
   ::= 10
-- Power Supply 2 failed
trap-ps2-failed TRAP-TYPE
   ENTERPRISE la5700
   DESCRIPTION
        "Trap, Power Supply 2 failed."
```

::= 11 -- Audio Loss (Primary) trap-pri-audio-loss TRAP-TYPE ENTERPRISE la5700 DESCRIPTION "Trap, Primary Audio lost or not detected." ::= 12 -- Audio Detected (Primary) trap-pri-audio-detected TRAP-TYPE ENTERPRISE 1a5700 DESCRIPTION "Trap, Primary Audio detected." ::= 13 -- GPO1 Turned On trap-gpol-on TRAP-TYPE ENTERPRISE 1a5700 VARIABLES {gpol-on-assignment} DESCRIPTION "Trap, GPO1 On" ::= 14 -- GPO1 Turned Off trap-gpol-off TRAP-TYPE ENTERPRISE la5700 VARIABLES {gpol-off-assignment} DESCRIPTION "Trap, GPO1 Off" ::= 15 -- GPO2 Turned On trap-gpo2-on TRAP-TYPE ENTERPRISE la5700 VARIABLES {gpo2-on-assignment} DESCRIPTION "Trap, GPO2 On" ::= 16 -- GPO2 Turned Off trap-gpo2-off TRAP-TYPE ENTERPRISE 1a5700 VARIABLES {gpo2-off-assignment} DESCRIPTION "Trap, GPO2 Off" ::= 17 -- GPO3 Turned On trap-gpo3-on TRAP-TYPE ENTERPRISE 1a5700 VARIABLES {gpo3-on-assignment} DESCRIPTION "Trap, GPO3 On" ::= 18 -- GPO3 Turned Off trap-gpo3-off TRAP-TYPE ENTERPRISE la5700 VARIABLES {gpo3-off-assignment} DESCRIPTION "Trap, GPO3 Off" ::= 19 -- GPO4 Turned On trap-gpo4-on TRAP-TYPE ENTERPRISE 1a5700 VARIABLES {gpo4-on-assignment} DESCRIPTION "Trap, GPO4 On" ::= 20

```
-- GPO4 Turned Off
trap-gpo4-off TRAP-TYPE
ENTERPRISE la5700
   VARIABLES {gpo4-off-assignment}
   DESCRIPTION
       "Trap, GPO4 Off"
   ::= 21
-- Codec Failed
trap-codec-failed
                     TRAP-TYPE
   ENTERPRISE 1a5700
   DESCRIPTION
      "Trap, Encoder/Decoder failed or not detected."
   ::= 22
-- Codec Detected
trap-codec-detected TRAP-TYPE
   ENTERPRISE la5700
   DESCRIPTION
        "Trap, Encoder/Decoder detected."
   ::= 23
-- Loudness Log record
trap-log-record TRAP-TYPE
   ENTERPRISE la5700
   VARIABLES {loudness-log-rec}
   DESCRIPTION
        "Trap, Loudness Log record."
    ::= 24
END
```

# 5.2 Logging

Using SNMP, the LQ-1 can report useful statistics that are captured via the status of the four GPOs. These are in addition to power supply health, audio presence, and Dolby decoder health. All of these can also be polled via GET commands, so it is possible, for example, to get loudness data more frequently if desired.

### **Trap Receiver**

Trap Receiver is a free and very useful application which runs on Windows XP through Windows 7. Assuming that you don't already have SNMP Management software running, it is probably the easiest way to collect traps sent out by SNMP-enabled units.

Visit www.trapreceiver.com to download the application. Install it by double-clicking. You may want to print a copy of the manual for reference as this tool is very flexible and has many options.

Once installed, configure it as follows:

- Click on the Trap Receiver Icon on your desktop.
- Click the Configure button in the lower right.
- Click the Logging tab.
  - Move the Flush Interval slider all the way to the left to flush often (1 Minute).
  - Pick a place where log files should go in the Filename: text box.
  - Click the Append radio button so that if Trap Receiver stops/restarts the log file won't be deleted.
  - We recommend changing the format to %i%g%s%bD
  - If you want to create log files, click the Turn Logging On radio button.
- Click the Miscellaneous Tab.
  - Pick a Max log file size. After the data in a log file reaches approximately this size, the old log file is closed and a new file with the next higher numeric suffix is opened. For example, if the original file name is mydata.log, after reaching this threshold that file will be closed and mydata001.log will be opened and used then followed by myda-ta002.log, etc.
  - Assuming that you are logging Linear Acoustic LQ-1 log data, and that you picked to log all parameters every 1 second (the maximum data at the fastest rate), the Max log filesize should be set for 11 Mbytes. This assumes that you will want each log file to

hold approximately 24 hours of data

• Click the Apply button to save your changes.

#### Notes

- When the Trap Receiver window is open, you will see any SNMP trap data sent to your PC. You do NOT need to keep this window open to receive the trap data into your PC and log file. Trap Receiver runs as a service behind the scenes, logging the trap data (if you set it to do logging).
- The Trap Receiver service will start running automatically when your PC boots up.
- If you don't see any traps, check:
  - Is the Linear Acoustic unit on the same LAN as your PC?
  - Can you ping the Linear Acoustic unit from your PC?
  - Is the PC's IP address entered into the Linear Acoustic unit? (Go to Setup -> Communication -> SNMP Trap Addr 1 to verify).
  - Is there a trap event happening at the unit?
- For LQ-1, turn on logging in Setup -> Meter Setup ->Log Frequency and set it to something other than "No logging".
- To test, set a GPO action that you can cause to occur. For example, go to Setup -> I/O -> GPO 1 Alert When. Set this to "Reference Loss," then remove and re-insert the reference to cause the alert to happen.
- Trap Receiver monitors port 162 (the standard SNMP Trap port) and Linear Acoustic units send traps to this port. If you have other SNMP Manager software running on the same PC, it will block Trap receiver from getting the trap data. You will need to stop the other software.

# Chapter 6: Troubleshooting

LQ-1 is reliable and easy to install. Most problems turn out to be wiring mistakes that apply incorrect input signals. However, it is a powerful device with many user-adjustable parameters, some of which interact. If you cannot find the source of a problem, simplify the signal path.

To instantly remove LQ-1 from the digital signal path, remove AC power from both inlets.

### The unit does not power on

1. Check whether the front panel logo is illuminated blue when power is applied to the unit.

The unit will then boot which will take about 30 seconds. The OLED screen will show activity. During that time you may hear the bypass relays cycle.

If the unit boots to the main screen, you are done.

If the lights are on but the unit still does not boot, continue.

- 2. Remove the AC cords and wait two minutes.
- **3.** Plug the unit back in.
- 4. The LQ-1 will automatically power back on.

### Output audio clicks and pops

The AES Reference may be missing or at the wrong sample rate; the unit expects to lock to 48 kHz. LQ-1 defaults to an internal 48 kHz reference if the external reference is removed or missing. This allows audio to continue, but with asynchronous inputs and outputs (due to the sample rate converters on each input pair). Make sure to connect a valid reference signal to avoid this issue.

An improper reference signal in equipment located up- or downstream from LQ-1 can also cause output noise. Use the master bypass function to verify that LQ-1 is not the problem.

### Input audio has dropouts

Clocking can be the root of all evil! Make sure it is set appropriately. Also, if monitoring through the headphones, note that there might actually be dropouts in the audio signal - the LQ-1 is a good test tool to determine this.

#### Audio issues disappear in bypass

Since the LQ-1 contains bypass relays, this can be a good way to isolate issues. Note that the LQ-1 does not process any audio, so unless the clock is set to an inappropriate setting (such as "Internal" when the AES output is connected into a digital plant), it may be that bypass is causing downstream gear to re-lock.

### Audio fixed by re-boot

Re-booting sometimes fixes audio problems by interrupting the AES signals, thereby causing equipment downstream to re-lock to the incoming signals. Selecting a preset with minimal processing (i.e., Protection Limit) may shed further light on the problem.

# **Chapter 7: Specifications**

Sampling Rate	48 kHz (±0.1%)
Output Delay	PCM, Dolby Digital, Dolby Digital Plus - 32 msec PCM/Dolby E - 33msec (NTSC) and 40 msec (PAL).
Program Configuration	Codec and metadata dependent; All standard modes supported.
Audio Word Length	24-bits
Metadata Input/Output	RS-485, 9-pin female D-connector on rear panel
GPIO Port	TTL level, 25-pin female D-connector
Ethernet Port	RJ-45 female jack connector
Headphone Output	
Frequency Response	20 Hz–20 kHz, ±0.5 dB
Distortion	Less than 0.2%, 20 Hz–20 kHz
Level	+12 dBu Maximum
Output Connector	1/4-in (6.35-mm) front panel jack
Digital I/O	
Digital Audio Inputs	Four unbalanced female BNC connectors, comply with AES-3ID-2001/ SMPTE 276M. Internal 75- $\Omega$ termination.
Digital Audio Outputs	One unbalanced female BNC connectors that comply with AES-3ID-2001/SMPTE 276M specifications.
LTC Input	Unbalanced female BNC connector per SMPTE 12M-1999. Supports 23.98, 24, 25, 29.97 and 30 Hz frame rates, including drop- and non-drop-frame modes
SDI I/O	
HD/SD-SDI I/O	Up to 16 channels of audio, signals per SMPTE 292/299M-2004, VANC metadata per SMPTE 2020 A or B.

Table 7-1 Electrical Specifications

Audio Decoder Types	Dolby Digital (AC-3), Dolby Digital Plus (E-AC-3) and Dolby E
Decoder Modes	<b>Dolby Digital</b> : up to 5.1 channels <b>Dolby Digital Plus</b> : up to 8 channels <b>Dolby E</b> : up to 8 channels plus metadata
Decoder Resolution	Up to 24-bit
Encoder Latency	Dolby Digital/Plus: 32 ms Dolby E: 33 ms NTSC, 40 ms PAL

Table 7-2	Audio	Decoding	Specifications
-----------	-------	----------	----------------

Table 7-3 Mechanical Specifications

Dimensions	1.75 × 19 × 11.5 in (44 × 483 × 293 mm)
Net Weight	6 lb (2.7 kg) approximate
Shipping Weight	8 lb (3.6 kg) approximate
Power Requirements	90-264 VAC, auto-sensing, 50-60 Hz
Power Consumption	35 W maximum

Table 7-4 Environmental	Spe	cificat	ions
-------------------------	-----	---------	------

Operating Temperature	0–50°C, convection cooled
Non Operating Temperature (Storage)	-20°C to 70°C
Humidity	Up to 98% relative humidity, non-condensing
	North America: Tested to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (CFR). Power supplies are UL tested and approved.
EMC/Regulatory	Europe: Tested to comply with the requirements of harmonised Low Voltage Directive 2006/95/EC and EMC Directive 2004/108/EC as indicated by the affixed CE marking; RoHS and WEEE compliant.

### Metadata Input Port

9-pin female D-connector with full-duplex RS-485 protocol running at 115 kbps. Pinout is compatible with SMPTE 207M. Pin-for-pin compatible with Dolby metadata sources (i.e. straight-through cable should be used.

Pin	Connection
1	Shield
2	TX A asynchronous data out –
3	RX B asynchronous data in +
4	Ground
5	NC
6	Ground
7	TX B asynchronous data out +
8	RX A asynchronous data in –
9	Shield

## Table 7-5 Metadata I/O Port Pinout

## **Ethernet Port**

Standard RJ-45 female connector that supports 10/100BASE-T.

## **GPI/O** Parallel Control Port

TTL level controls, active Low.

Pin	Function	Pin	Function
1	GPI 1	6	GPO 1 (150 mA)
2	GPI 2	7	GPO 2 (150 mA)
3	GPI 3	8	GPO 3 (150 mA)
4	GPI 4	9	GPO 4 (150 mA)
5	Ground		

Table 7-6	GPI/C	) Parallel	Control	Port	Pinout
Table $7-0$	UL1/C	) raraner	Control	FOIL	rmout