Dolby

Model 737 Soundtrack Loudness Meter - Leq(m)

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Dolby Model 737 Soundtrack Loudness Meter - Leq(m)

User's Manual

Software Version 1.0

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Chapter 1
Introduction

Leq(m) is a term used by Dolby Laboratories to describe the level of annoyance in movie soundtracks. The Leq(m) value is a weighted true average of the audio power level sent to the camera in optical or digital soundtracks, or the same weighted true average of the audio power level passing through a cinema processor before equalization. The Dolby **Model 737 Soundtrack Loudness Meter - Leq(m)** provides a convenient method of measuring soundtrack annoyance.

The phrase *true average* signifies that the meter measures the power at regular intervals, adds these quantities together and then divides the result by the length of the measurement time. Since the measurement time is a critical factor in arriving at an accurate result, the measurement should start and stop at known points. These points can range from the *First Frame of Action* (FFOA) and *Last Frame of Action* (LFOA) of a spot such as a ten second clip, to the beginning and end of an entire movie. The degree of repeatability of Model 737 measurements is dependent on the length of the measurement. For example, a ten second spot requires about two frame accuracy whereas a two minute thirty second trailer needs only two second accuracy for correct measurement. For an entire feature, the measurement can be started and stopped at a convenient time, within two minutes of the opening and closing credits, to get repeatable results. Measurements made within 1% of the stated time frame will compare accurately.

The Model 737 consists of six input channels with setup trims followed by five weighting filters, a true averaging power meter that can be remotely started and stopped, and a trip circuit that indicates a preset maximum level has been exceeded. The unit is intended for use on soundstages, optical recording facilities, and film QA installations to audition the final two- and six-channel mixes that will be recorded on the film. It can also be used in theatres to check levels during exhibition.

NOTE: *See* **Appendix A Background** *for a complete discussion of the factors that led to the creation of the* Model 737 Soundtrack Loudness Meter - Leq(m).

Chapter 2
Operation and Features

The **Model 737 Soundtrack Loudness Meter - Leq(m)** is a simple 1U, rack-mountable device that measures the equivalent loudness of one-, two-, four-, or six-channel inputs, typically from the non-equalized outputs of a cinema processor or mixing console. Since the time available for averaging is several hours, a soundtrack can be evaluated in small sections or in its entirety. The current Leq(m) value is displayed in the LED window. The circuit consists of a set of six input stages, five CCIR (2K) filters, five square law detectors, and a display driver that takes the average of the squared signal voltage and then extracts the square root/log of the resultant DC signal. A separately processed average DC circuit outputs a signal that is buffered for use with a chart recorder/ moving coil meter for trend analysis.

The **Trip Set** adjustment sets the level that lights the **Trip LED** and triggers a signal to the **Trip Output**. The Trip Output can activate a more noticeable device (such as a light or sound) to alert the user that a preset level has been exceeded at the end of the measurement.

2.1 Front Panel Controls

The front panel has three pushbuttons (Cal, Start, Stop), a two-digit LED that displays the current Leq(m) value, a screwdriver-accessible hex switch to set the trip level, trimpots for the six analog inputs, and a **6/4 – 2** channel switch.

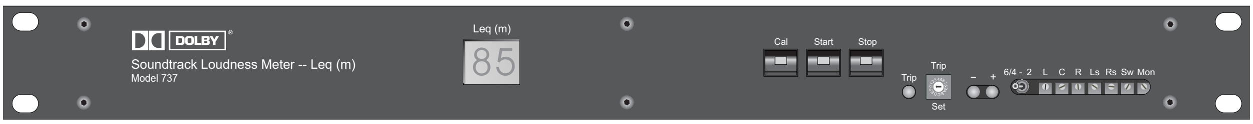

Figure 2-1 Model 737 front panel

The right side of the front panel is shown in detail below.

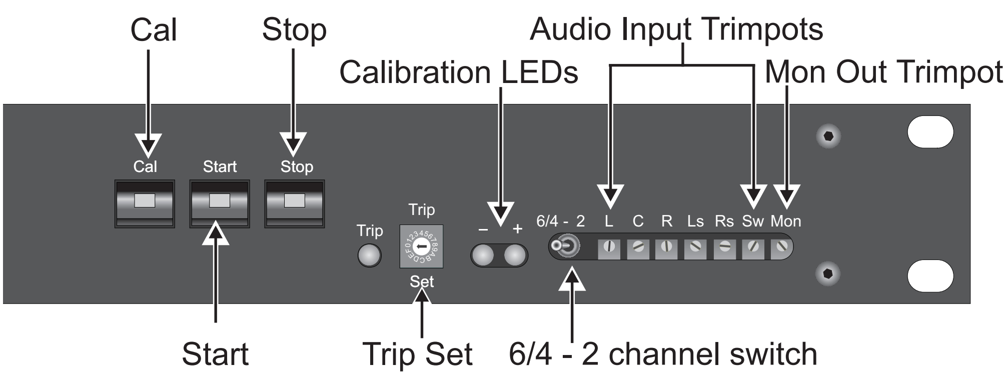

Figure 2-2 Detail of front panel controls

2.1.1 6/4 – 2 Channel Switch

Located to the left of the input trimpots (see Figure 2-2), this switch may be used in future versions to introduce a fixed offset in the measurement of two-channel encoded material. Presently (at Version 1), the offset is set to zero so *the switch has no effect on the measurement*.

2.1.2 Input Trimpots

To derive accurate and repeatable Leq(m) measurements, the input trimpots are used to calibrate the Model 737 to standard input levels. The input trimpots are screwdriver-adjustable and allow sensitivity adjustments from -12 to +4 dBu for Dolby Level to suit a wide variety of console setup levels and cinema processor types. See section 2.3 for a complete discussion of how to connect and calibrate the Model 737 to different signal sources.

2.1.3 Trip Set

Trip Set is a 16-position switch that sets a value from 78 Leq(m) at position *0* to 92 Leq(m) at position *E* in 1 unit increments. When **Stop** is pressed to end the measurement, if the Trip Set level has been exceeded, the Trip LED lights and a signal is sent to Trip Out. The Trip LED does not light during the measurement to allow transient Leq(m) values to exceed the Trip Set value without triggering Trip Out.

NOTE: *The Trip circuit can be disabled by setting the switch to position F.*

The current Trip level can be checked by holding down the **Cal** button then pressing and holding the **Stop** button. This test always sends a signal to **Trip Out** to check the circuit operation. Push the **Run** button to cancel the trip indication. In position *F*, the display is blank.

2.1.4 Pushbuttons

- **Cal –** Push to enter calibration mode to set audio input levels using the trimpots. In calibration mode, Mon Out sends a mono mix of the channels instead of the output of the filter. Push Start to return to measurement mode.
- **Start –** Push to begin a measurement.
- **Stop –** Push to stop a measurement and display the final level of the measurement.

2.2 Back Panel Connectors

The back panel has six audio input connectors (shown in detail in Figure 2-4), two output connectors and a power/control signal connector (shown in detail in Figure 2-5).

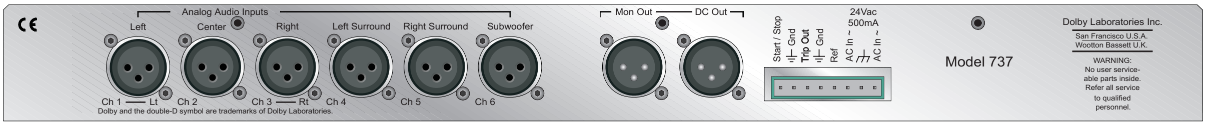

Figure 2-3 Model 737 back panel

2.2.1 Analog Audio Inputs

Although the input stages of Left, Center, Right, and Left Surround are identical, it is recommended that Left and Right (Ch 1 and Ch 3, also labeled Lt and Rt, respectively) be used for Lt/Rt measurements of two-channel encoded material. The Right Surround and Subwoofer channels share a common filter and rectifier so the Subwoofer input is not normally used except when measuring 5.1-channel material.

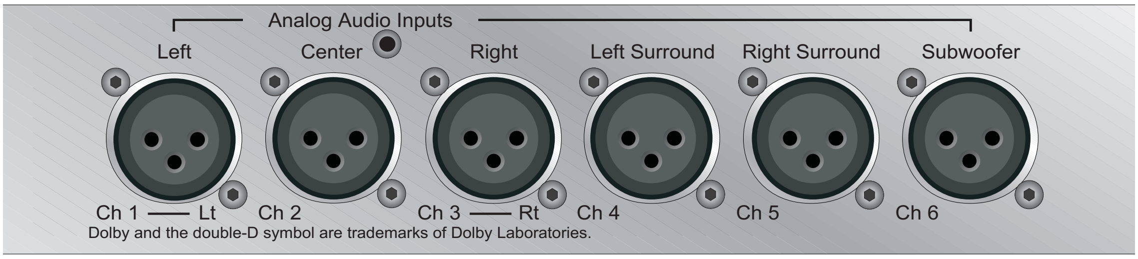

Figure 2-4 Audio input connectors

The XLR input connectors for channels 1 – 6 use the following wiring convention:

Pin	Connection
1	Chassis
2	Audio +
3	Audio -

2.2.2 Mon Out

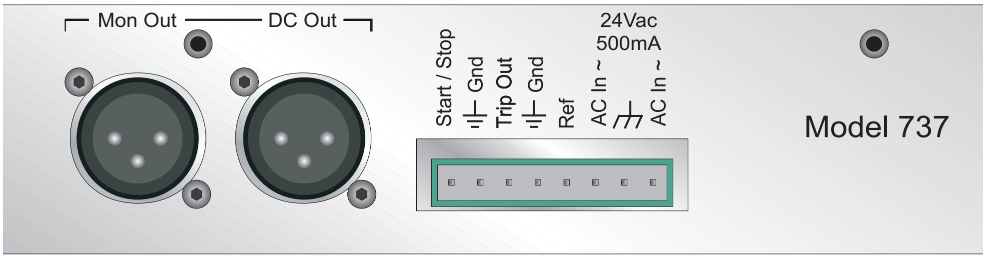

Figure 2-5 Mon and DC Out, and power/control connector

Mon Out is an unbalanced output which, in calibration mode, provides an unweighted mono mix of channels. In measurement mode, Mon Out carries a mono mix of the outputs of the weighting filters. Listening to the filtered output can help the mixer find the effect or equalizer setting that is raising the average above the standard. Mon Out is adjustable on the front panel from -20 to 0 dBu for Dolby Level so that it can be balanced with the studio monitor level.

Mon Out uses the following wiring convention:

Pin	Connection
1	Chassis
2	Mon signal out
3	Mon signal common

2.2.3 DC Out

DC Out is an unbalanced output to drive an optional moving coil meter/chart recorder. Using a 1 mA moving coil meter with a calibrating potentiometer across its terminals is a useful method to observe measurement trends. DC Out uses the following wiring convention:

Pin	Connection
1	Chassis
2	DC out
3	DC return

1 V is equivalent to 91 Leq(m), the range is 0 – 1.25 VDC, 0 VDC corresponds to 70 Leq(m), output resistance is 1 kΩ.

2.2.4 Power and Control

The 8-pin power and control connector uses the following wiring conventions for the control and AC power signals:

Pin	Connection
1	Close to stop (+)
2	Close to stop (gnd)
3	Trip indicator (+)
4	Trip indicator (gnd)
5	Vref
6	24 VAC
7	Chassis ground
8	24 VAC

The trip output is a simple open collector buffer that passes current when the trip value is exceeded. The voltage on the open trip contacts should not exceed 24 VDC and the current passed when closed should not exceed 20 mA.

2.3 Connection and Calibration

The Model 737 may derive its input from a variety of signal sources including an audio console, a cinema processor, the SDU4 Surround Decoder, the DA10/20 Digital Adapters, and non-Dolby digital adapters. The Model 737 accurately measures both matrix-encoded two-channel and decoded four- and six-channel sources. The following three important points are independent of the signal source:

- Dolby Level should be used as the test signal input level. In the digital domain, Dolby Level correlates to -20 dBFS (20 dB below **F**ull-**S**cale digital).
- The calibration for each input must be performed with *only* that input connected. When that input is calibrated, unplug it and plug in the next input to calibrate. All inputs are connected *after* the calibration in preparation for the actual measurement.
- Matrix-encoded two-channel measurements are normally 3 - 5 dB lower than the same signal measured in a decoded format.

The calibration procedure is discussed in section 2.3.6. The following sections discuss how to interface the Model 737 to the most commonly used signal sources.

2.3.1 Connecting to an Audio Console

Connect the Left, Center, Right, Left Surround, Right Surround, and Subwoofer outputs from the audio console into the appropriate inputs on the Model 737. Use a 1 kHz tone at Dolby Level as the test signal to each input.

2.3.2 Connecting to a Dolby Cinema Processor

The method for connecting a Dolby cinema processor with the Model 737 depends on the application. For a cinema processor used in a studio or dubbing stage, the matrix-encoded Lt and Rt signals are found on the Cat. No. 150E card. If the connection for measurement is occasional, use the red and green test points, respectively. For more permanent installations, connect to pins L and J on the rear panel of the cinema processor.

For a cinema processor used in a theatre, the Lt and Rt signals are found on the red and green test points (ground is black), respectively, on the Cat. No. 150E or Cat. No. 150F. The Cat. No. 150D does not have test points. Thread up and play a length of Cat. No. 69T test film at Dolby Level as the signal source. Set the unit to Format 04 for the calibration steps that follow in section **2.3.6 Calibration Procedure**.

NOTE: *Alignment in Format 05 or mono will result in incorrect levels.*

2.3.3 Connecting to a DA10/20 Digital Adapter

When checking levels in a theatre or print-checking facility, the Model 737 will often derive its input signals directly from the J8 connectors on the back of the DA10/20. The following table lists the CP Audio Out Conn. J8 pin number signal assignments and the nearest ground pin.

Table 1 DA10/20 Audio Out Conn. J8 Signal Assignments

Output	Active	Ground
Left	14	9
Center	20	12
Right	17	11
Left Surround	15	10
Right Surround	2	1
Subwoofer	24	13

For each channel, the active pin of the output is wired to pin 2 and the ground to pin 3 of that channel's XLR connector. If the units are in the same rack, there should be a connection between the metal parts of the Model 737 and the digital adapter. If the Model 737 is not in the same rack, there must be a connection between any of the ground pins (1, 4 - 13) of J8 and one of the ground pins on the 8-pin power and control connector on the rear panel of the Model 737.

When used in a theatre to check released prints, thread up and play a length of Cat. No. 69T (US) or Cat. No. 1010 SR.D test film at Dolby Level as the signal source.

2.3.4 Connecting to Non-Dolby Digital Adapters

With the digital adapter connected to the Dolby cinema processor, make a parallel connection from the cinema processor to the audio inputs on the Model 737, and use the test media from that manufacturer as the signal source to apply Dolby Level to each channel.

2.3.5 Connecting to the SDU4 Surround Decoder

The Model 737 can interface with the SDU4 either before or after decoding. To measure the matrix-encoded signal, connect the signals intended for the SDU4's Lt and Rt inputs to the Lt and Rt inputs (channels 1 and 3) on the Model 737.

To measure the decoded signal, connect the four-channel L, C, R, S outputs from the SDU4 to the Left, Center, Right, and Left Surround inputs on the Model 737.

NOTE: *The matrix-encoded signal will measure 3 - 5 dB lower than the decoded signal.*

2.3.6 Calibration Procedure

To obtain an accurate Leq(m) measurement, the input trimpots must be used to calibrate the inputs to standard levels to account for the differing levels and formats of the various signal sources. Follow these steps to calibrate the inputs:

1. Press the **Cal** button.
2. Send the appropriate test signal at Dolby Level to the Left input only.
3. Adjust the input trimpot associated with that input until the + and - LEDs beside the trims are of approximately equal brightness and the Leq(m) value displayed is 85.

   The window of equal illumination is designed to be extremely narrow, which yields an accurate level setting when both LEDs are on, albeit dimly (the gain error is less than 0.1 dB).
4. Unplug the Left input and repeat steps 2 – 4 for the:
   - Right and Center inputs for 5.1-channel material.
   - Right input for matrix-encoded material.
   - Right and Center inputs for four-channel (L, C, R, S) soundtracks.

The next steps apply to soundtracks with at least one surround channel.

5. Send the appropriate test signal at Dolby Level to the Left Surround only.
6. Adjust the trim associated with that input until the + and - LEDs beside the trims are of approximately equal brightness and the Leq(m) value displayed is:
   - 82 for soundtracks with stereo surround channels.
   - 85 for soundtracks with one mono surround channel.
7. For soundtracks with stereo surrounds, unplug the Left Surround and plug in the Right Surround input and repeat steps 5 – 6.

The next step applies to those soundtracks with a Subwoofer channel:

8. Change the tone frequency to 100 Hz at Dolby Level and set the Subwoofer trim to an Leq(m) value of 85.

As a final check, plug in all inputs and send the appropriate tone to all applicable channels. The meter should register an Leq(m) value of 91 for a 5.1-channel soundtrack and 88 for a two-channel soundtrack. Repeat the calibration if there is a significant discrepancy.

NOTE: *The 3 dB difference between the L,C, R and surround channels matches the transfer levels for SR.D movie soundtracks. The null circuit has two points of transition (at 82 and 85) so the + and - LEDs should be used in conjunction with the numerical display to set the correct levels for that input.*

2.4 Mains Power

The Model 737 uses a mains transformer that converts incoming AC to 24 VAC. The transformer has 1 meter leads and should be placed at the base of the equipment rack in which the unit is mounted. The transformer should not be operated with the 24 V leads shorted. Transformer units are available for 200 – 240 VAC input and 100 – 125 VAC input. Please ensure that the correct transformer is ordered for your local mains voltage.

- For 110 VAC, Dolby part number is 54058
- For 230 VAC, Dolby part number is 54057

In some countries, the primary cable for the module may not have a mains plug fitted. These unterminated leads must be properly wired to a mains plug in accordance with the following international code:

Brown wire: Live or hot

Blue wire: Neutral

NOTE: *If you are uncertain about the wiring of your mains outlet do not use it. Consult a qualified electrician.*

2.5 Interface

The connections for remote start/stop control are normally wired to a relay across the projector motor or to a console mounted toggle switch. When the contacts close or the **Stop** button is pressed, the display flashes the last measured value. When the contacts open again, or the **Start** button is pressed, the unit enters measurement mode. Pressing the **Cal** button causes the unit to enter calibration mode regardless of the last function pressed. To remotely start a measurement from calibration mode, the contacts must close and then open again. The contacts should be capable of passing 10 mA and should not have contact with other circuits. A simple toggle switch or a pair of uncommitted relay contacts is ideal.

2.5.1 Grounding

There is no connection between the audio common inside the unit and the metalwork. For most grounding schemes, a link of 16 AWG between pin 4 and pin 7 of the power/control connector provides a satisfactory grounding scheme. If this induces hum because of a loop via the rack mounting, the link can be removed.

2.6 Block diagram

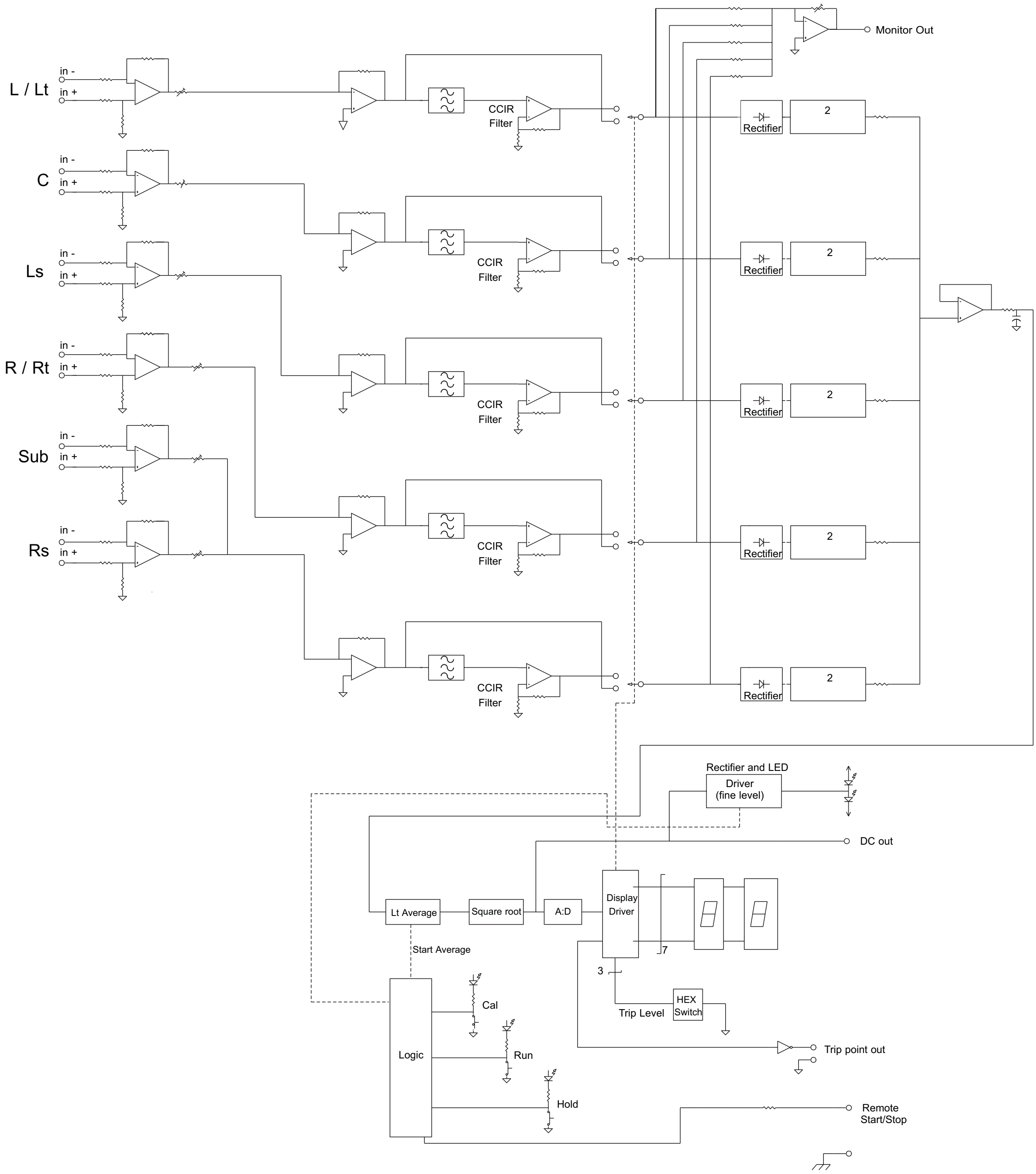

Figure 2-6 Model 737 block diagram

The six input channels each have a balanced to unbalanced stage buffering the input signal. Trimpots on the front panel set each channel's input level. The unbalanced signals then pass to the metering CCIR filters and make-up gain stages. The signals on the "wipers" of the selector switch, controlled by the Mon trimpot, are buffered and sent to Mon Out on the rear panel.

The selection switches are followed by five ideal rectifiers and squaring circuits. The five squared signals are added and short term averaged. The resultant varying DC signal passes to an analog long-term averaging circuit. This requires a logic signal from the pushbutton logic section to start and stop the average measurement. The analog long-term averaged DC signal is sent to the DC Out connector on the rear panel for optional connection to a chart recorder or moving coil meter.

The short-term average DC voltage is converted to12-bit binary and sent to a 6805 processor that also averages but over a much longer time span. The samples are taken at 850 msec intervals, summed, divided by the time since the Start button was pressed, and then converted to a square-root value. This value is then converted to a dB (log) scale, scaled to a reference level of 85, and sent to the front panel LED display. When the **Stop** button is pressed, the averaging circuit and averaging routine are reset while the last value measured is displayed.

The pushbutton logic is a simple tristable gate that drives the LEDs in the pushbuttons and also generates processor interrupts to indicate mode changes. It is remoted to the back panel via a single start-stop line.

Chapter 3
Specifications

Temperature and Humidity	
Operating	10^{o} C to 35^{o} C, natural convection cooling
Non-Operating (Storage)	0^{o} C to 70^{o} C
Humidity	Up to 90% relative humidity, non-condensing
EMC Limits	
AC power line conducted	Per EN 55022
Radiated	Per EN 55022
Immunity	Per EN 50082-1
General	
Dimensions	43 x 20 x 4.5 cm (19 x 7.8 x 1.75 inches)
Net Weight	3 kg (6 lbs approx.)
Power Requirements	24 VAC, 500 mA wall transformer (provided)
Power Consumption	10 WAC max.

Appendix A
Background

This Appendix is an edited version of a paper published in the January 1998 *SMPTE Journal* (Volume 107, Number 1) entitled “Are Movies Too Loud” by Ioan Allen. The complete paper is available from the Dolby website at www.dolby.com/movies. Reprints of the paper are available by sending email to info@dolby.com (include the article title) or by telephone at 415-558-0200.

Over the past few years, the film production community has become increasingly concerned that movies are getting louder. In addition, movie-goers often complain that movies are too loud and, as a result, many theatres now reduce the audio fader below the calibration level used in the dubbing theatre.

This chapter introduces some of the issues that motivated the creation of the Dolby **Model 737 Soundtrack Loudness Meter - Leq(m)**:

- What factors affect soundtrack “loudness?”
- How can loudness be quantified?
- Do new sound formats (Dolby SR, Dolby SR·D, Sony SDDS and DTS) exacerbate the problem?

A.1 Loudness

Those familiar with movie soundtracks won’t be surprised at the subjective statement that *The Right Stuff* (1983; Dolby A-type 70 mm and Dolby A-type stereo optical) was a loud movie. *The Right Stuff* was a subjectively louder movie than *Shine* (1996, Dolby Digital). Since *Shine* was subjectively louder than *Days of Heaven* (1978), perhaps a case could be made that movies have been getting both louder and quieter! Of course, this does not lead to an objective assessment; selected titles can be used to prove either trend. *Gone with the Wind* (1939) could be used to argue that films have become quieter over the last six decades!

A.1.1 The Current Situation

In the mid-1970s, Dolby Laboratories introduced a calibration recommendation for monitor levels in movie soundtracks. A pink noise reference signal was used in the record chain to adjust the audio monitor level to 85 dBc. All theatres equipped for playback of the new stereo optical soundtracks were set up such that an equivalent pink noise signal would generate the same 85 dBc with the playback fader set to the calibrated setting. This meant that theatres playing films at the calibrated fader setting (fader 7 on most cinema processors) would reproduce the same volume level selected by the film director and audio engineers in the dubbing theatre.

This system worked quite well for many years. Dolby Stereo (A-type encoded) films had limited headroom and the resulting constrained dynamic range yielded few audience complaints. Most theatres played films at the calibrated level. Soundtrack format technology has been significantly enhanced since Dolby Stereo. Dolby SR extended the headroom by 3 dB at midrange frequencies, and more at low and high frequencies. In recent years, the new digital formats have further increased the headroom (Figure A-1).

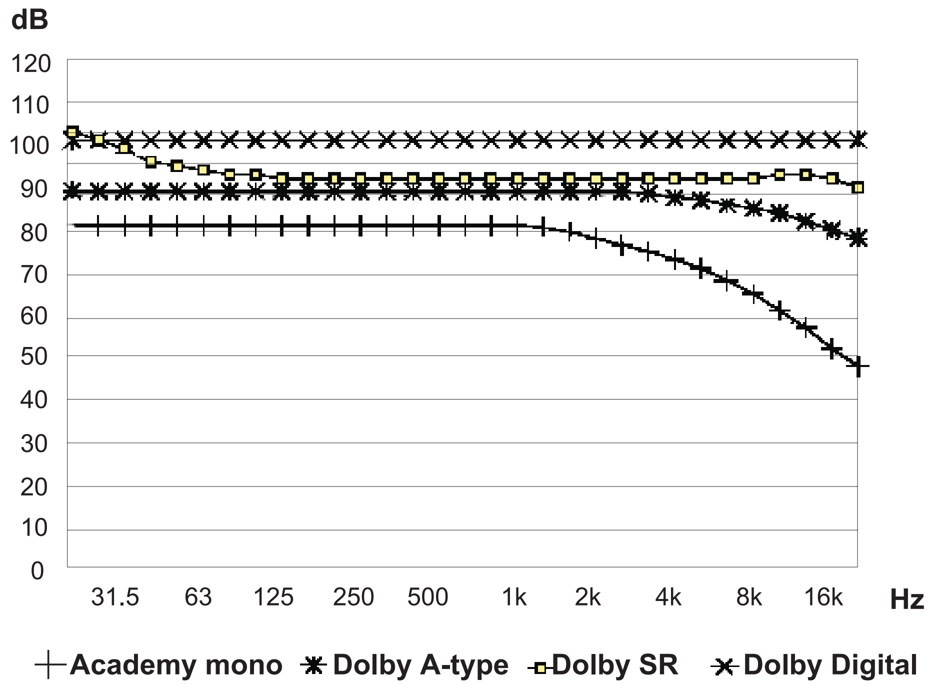

Figure A-1 Peak levels of photographic soundtrack formats.

Because the 85 dBc calibration technique has been maintained throughout these format changes, additional headroom is available on the newer soundtracks. Feature films have one consistent, subjective mix reference for dialogue record level, known as “associative loudness.” When the dubbing mixer sees an actor on the screen, and there is no “fight” with music or effects, the dialogue level in a moderate close-up is set to be plausible for the visual. Within reasonable limits, this holds true to within 2 or 3 dB. This natural dialogue level does not hold true for narration, as there is no corresponding visual reference. Music and effects have no direct visual associative loudness. Most people are not familiar with the actual sound pressure levels of a Concorde take-off or a 50 mm howitzer. The music score level is equally uncalibrated.

As the headroom capability of the recording medium has been extended, it has certainly been used: the “non-associative” loudness of effects and music has risen to fill the available space. The discretionary use of this increased headroom would be justified and desirable on some feature films, such as “ride” and action movies.

In practice, the following undesirable symptoms have arisen:

- There are an increasing number of audience complaints that movies are too loud. Newspaper articles have been written on the subject and respected sound mixers have spoken publicly about the problem.
- Theatres are playing films substantially below the calibrated level. A fader level of 5, as opposed to the calibrated level of 7, is not uncommon, representing a level reduction of approximately 6 dB.
- Trailers are fighting for competitive loudness. Theatre playback levels are often set by complaints generated by the loudest (and earliest) element of the show. If the playback level is set in response to the loudest trailer, the feature often plays at the same reduced level. The result is that the dialogue level of the feature is lowered by the same level deemed necessary to attenuate the trailer. A feature film played 6 dB below the calibrated level may have serious dialogue intelligibility problems.
- Mix engineers are using ear plugs to avoid the risk of hearing damage.
- In Europe, where commercials are played before the feature, competitive loudness has led to the desire for a uniform measurement technique, and a self-disciplined constraint.

It is possible that the increased use of headroom from Dolby A-type to Dolby SR and digital releases has not been matched by a corresponding increase in power amplifier and loudspeaker capability. The resultant distortion from overloaded equipment may well exacerbate the loudness problems of recent soundtracks, causing increased incidence of complaints.

A.2 How Loud Is a Movie?

Loudness is an extremely subjective term and has been defined and measured in many different ways. Various definitions have arisen from a desire to quantify loudness in specific situations:

- How annoying is the background noise level in a working space?
- How damaging is sustained, high-level noise exposure?
- How intrusive is the noise level of a recording or transmission medium?
- How can the instantaneous loudness of different spectra be compared?

In attempting to measure the loudness of a movie or trailer soundtrack, conventional level meters in the recording chain are of only marginal help. A VU meter has slow time constants, and is of little use in detecting short-term peaks. The PPM meter was designed to show short-term peak levels that might clip the recording or transmission media. Neither system demonstrates an index of what determines the *perceived* loudness of a film soundtrack.

None of the existing criteria for loudness can be directly applied to a soundtrack. Fundamental loudness is defined by the relationship between frequency and level. A regular VU or PPM meter does not account for the ear's varying sensitivity with respect to frequency. Sound level meters account for some frequency/level factors and are typically switchable to different *weightings*. A-weighting, for example, attempts to account for the ear's decreased sensitivity to low frequencies.

The length of the sound is another important factor in determining the perceived loudness of a sound. It is well known that people attending a loud three-hour rock concert may suffer from some temporary hearing loss after the concert. A five-minute exposure to the concert, however, does not create the same effect. It can be construed that the longer a loud sound lasts, the greater the apparent loudness or *annoyance*. Loud sounds of short duration may cause great surprise, but little annoyance. A sudden gunshot in the middle of an otherwise quiet scene causes few, if any, complaints. Repeated gunshots, however, can distress an audience.

One measure of sustained loudness is called *Leq* (loudness equivalent), which was originally derived to gauge potential hearing damage from exposure in industrial environments to sustained, varying-level sounds. Leq can be defined as the level of a steady-state tone with an equivalent level as the level of a time-variant signal. The original intent was to define potential hearing damage in industrial noise environments. Several different formulae define Leq, but all perform averaging of the level of material over time.

A.3 Standardizing a Measurement Technique for Soundtrack "Loudness"

The following factors must be considered to develop a valid measurement technique:

- **frequency weighting –** determining the frequency ranges that most closely correlate to loudness annoyance
- **long-term averaging –** measurement definitions such as Leq

It would be highly desirable to combine these concepts to produce a single value that accurately represents the loudness of a movie.

Dolby Laboratories decided to set up a variety of measurement techniques to evaluate film samples. Obviously, it was impossible to measure a totally comprehensive set of material. The selected samples used for testing included: contemporary digital trailers, sections of recent digital releases, maximum level Dolby SR and A-type releases, typical dialogue-only recordings in various formats (including Academy mono), and UK commercials.

A.3.1 Alternative Measurement Indices

As might be expected, the samples yielded different loudness values when assessed with different systems. The low-frequency roll-off of A-weighting results in a reduction in Leq when the material has a substantial bass content. Determination of annoyance, however, does not necessarily match the A-weighting curve. Research at Dolby has revealed that placing heavier emphasis on the 2 – 6 kHz region better matches how people react to soundtrack loudness. It was also discovered that the CCIR-weighting curve used to measure low-level recording medium noise more closely matched the subjective annoyance criteria (Figure A-2).

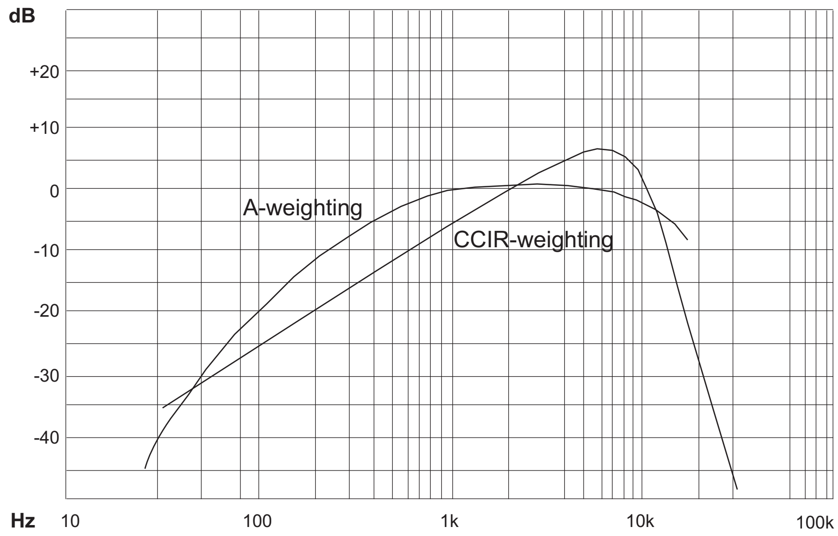

Figure A-2 A-weighting vs. CCIR-weighting curve (offset by 5.6 dB)

Although there is no technical parallel between high-level soundtrack loudness and low-level recording medium noise, the CCIR curve provided a convenient weighting filter for the tests and a better subjective match than Leq(a). The CCIR curve can be further adapted to represent soundtrack loudness by offsetting the level by 5.6 dB, with a 2 kHz reference point. Loudness values derived from this characteristic are referred to in this document as **Leq(m).** Figure A-3 shows the relationship between Leq(m) and Leq(a) for UK commercials. Although the *average* values for Leq(m) and Leq(a) are the same, higher relative Leq(m) values show a signal content with greater emphasis in the 2 – 6 kHz region.

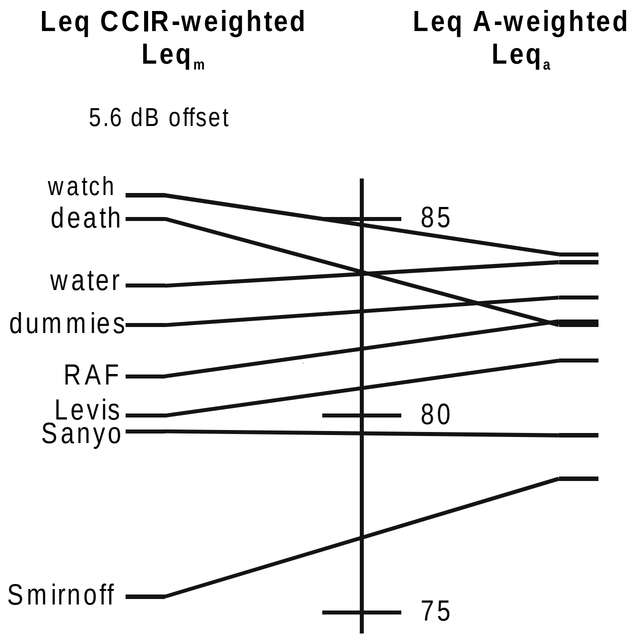

Figure A-3 CCIR-weighting (Leq(m)) shows influence of 2 – 6 kHz region.

A.4 Analysis of the Data

Figure A-4 shows the relationship between un-weighted (Leq) and CCIR-weighted (Leq(m)) samples. The hierarchy on the right of the figure (un-weighted) differs from that on the left (weighted). The slope of the connecting lines denotes the amount of bass content in the program. Lines with higher slopes indicate larger amounts of bass, while horizontal lines (0 slope), or lines with negative slope indicate less bass content. Of the samples shown, notice that *Indiana Jones and the Temple of Doom* is raised in the hierarchy due to the lack of bass in the five-minute sample. This matches the *apparent* loudness of the sample, one of the loudest Dolby A-type films.

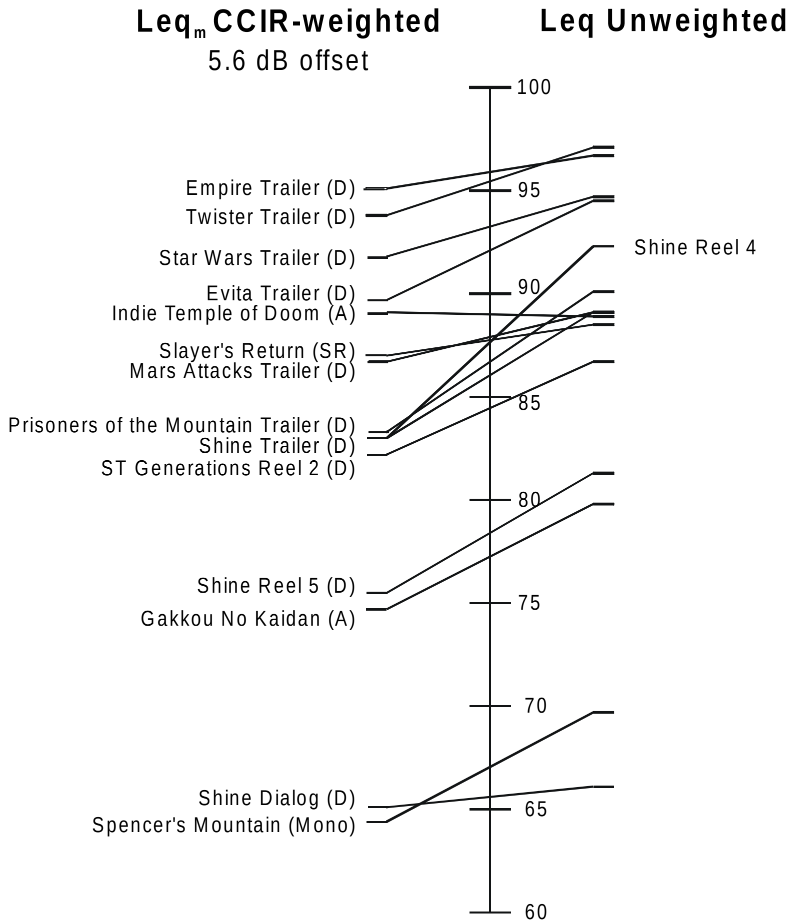

Figure A-4 Comparison of weighted and un-weighted Leq.

Figure A-5 compares the maximum peak level reached in each sample with the weighted Leq(m) value. The slope of the line between the values is now proportional to the *dynamic range*: the range between the average level and the loudest peaks on the soundtrack. It is important to consider the dialogue level in these samples. The concept of associative loudness (discussed above) is supported by the comparison of dialogue levels of *Spencer's Mountain* (1963), an Academy mono film, with the dialogue level of *Shine* (1996), a Dolby Digital release. The two films show identical Leq levels, confirming the supposition that mixers set dialogue levels at a plausible point (associative loudness), regardless of the format of the release.

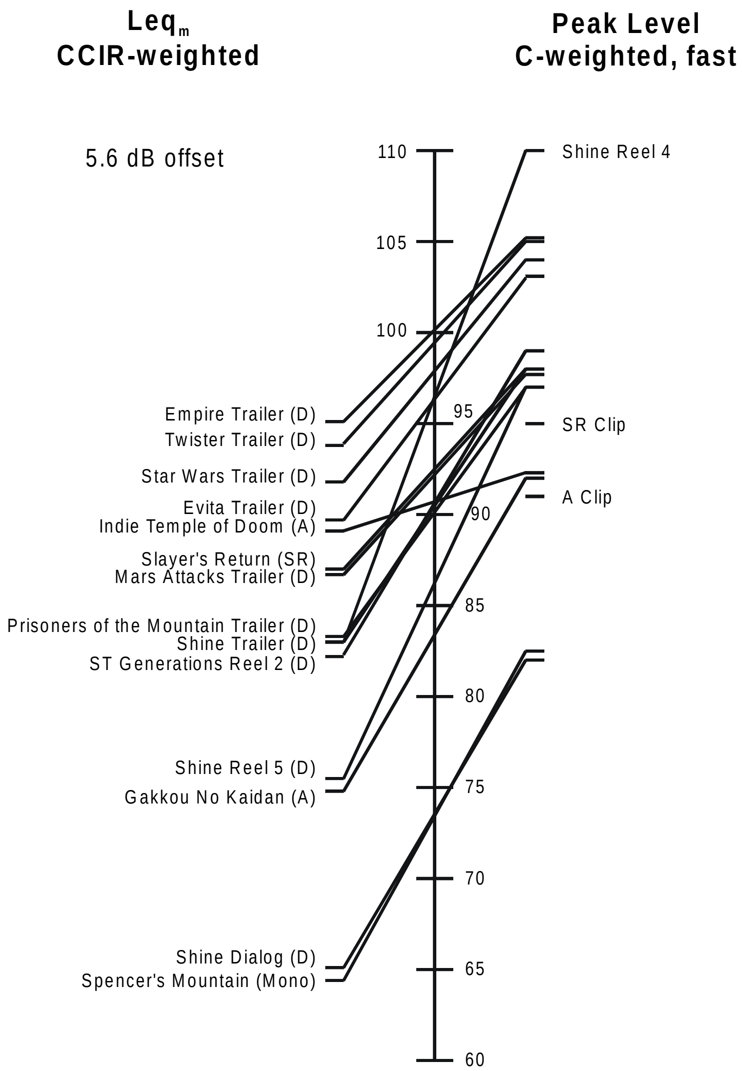

Figure A-5 Comparison of Leq(m) and peak levels

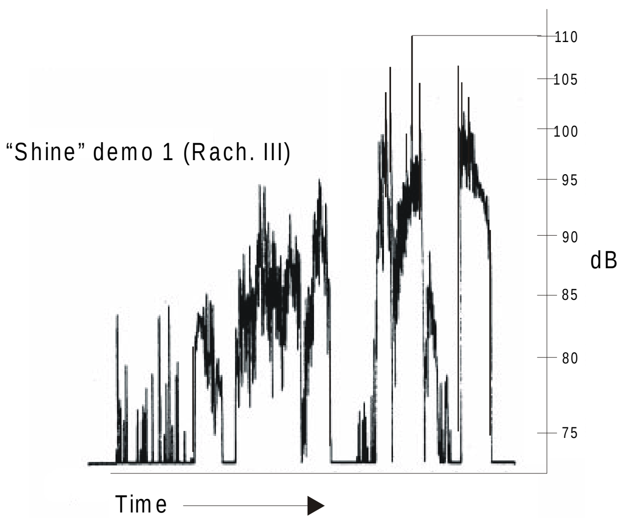

Figure A-6 Peak levels in reel 4 of Shine

Next, examine Figure A-5 again, then Figure A-6, and compare the peak levels in reel 4 of *Shine* with the Leq. Although this reel has an extreme dynamic range, audiences do not complain that it is too loud. In this case, the dynamic range is correct for the nature of the material, and the Leq is much more indicative of the subjective loudness than a peak measurement.

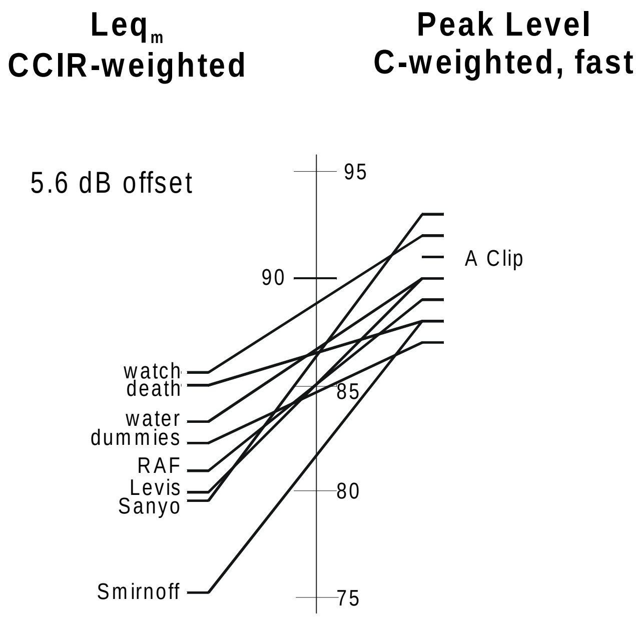

Figure A-7 Comparison of Leq(m) with peak levels for UK commercials

Figure A-7 shows that some of the UK commercials are pushing at the maximum available loudness of the media format. Even these Dolby A-type soundtracks show a loudness far greater than a constraint on maximum peaks would control. Attempts have been made in Europe to restrict loudness of commercials by defining a maximum level with respect to 100% of the medium's clipping level. For example, a 50% limit would mean that no peaks would be permitted to exceed a level 6 dB below the maximum excursion of a Dolby A-type stereo optical release. This constraint, however, would have the effect of restricting dynamic range, and not necessarily of controlling the subjective loudness of the material. US-created digital trailers typically have an Leq(m) level up to 15, or even 20 dB above the feature they advertise!

It must be emphasized, however, that the higher Leq(m) values of trailers is caused, in part, by the selection of only the loudest, most exciting segments of the movie packed into the 90 second or 2 minute sample. When the entire movie is averaged, the Leq(m) value is normally much lower.

A.4.1 Verification of the Data

Although the prime measurements for the data were direct electrical sums of the analog or digital film itself, peak levels were checked periodically with a sound pressure level meter in an auditorium. The lines indicating Dolby A-type and SR show the maximum theoretical levels of these formats. The data shows the maximum recorded levels are slightly higher than these theoretical numbers. This is not surprising and is due to the slight excessive dimensional modulation consistent with contemporary practice.