

REPORT ITU-R BS.2103-1

Short-term loudness metering

(Question ITU-R 2/6)

(2007-2008)

Foreword

This Report is in two parts. The first part discusses the need for different types of programme loudness meter for different contexts, discusses general psychoacoustic principles, specific psychoacoustic principles in the context of temporal loudness sensation, and discusses the incorporation of these principles into ballistic characteristics for short-term loudness meters. The second part of the Report deals with correlation of short-term meter characteristics with loudness listening tests. This correlation is examined for several possible equalization curves and for a range of ballistic rise and fall time constants.

The purpose of a short-term loudness meter in broadcasting is to give a short-term indication that reflects the likely contribution to long-term loudness, rather than short-term loudness per se. While these quantities are related, recent research has shown that they are not the same thing. The references used for performance assessment of each short-term meter variant are therefore listening test results on long-term loudness.

CONTENTS*Page*

Part 1

1	Introduction	2
2	Application specific loudness meters	2
3	Human loudness perception	3
4	Review of proposed ballistic types.....	4
5	Implementation – theory.....	6
5.1	Algorithm topology	6
5.2	Output filter topology	7
5.3	Time constant for instantaneous and fast response.....	7
5.4	Audio block decimation.....	7

	<i>Page</i>
6	Implementation – Loudness meter comparison utility 8
6.1	Introduction..... 8
6.2	Calibration 8
6.3	Decimation..... 8
6.4	Gating 10
7	Conclusions 10
Part 2	
1	Purpose 11
2	Method..... 11
3	Results 12
3.1	Meter types 12
3.2	VU and PPM results 13
3.3	Loudness meter results 15
4	Conclusions 19
5	References and Bibliography..... 20
	Appendix 1 – Results in numerical form 21

Part 1

1 Introduction

Radiocommunication Study Group 6 has approved a number of new Recommendations for international standardization of loudness meters. This has occurred in response to a need by many member administrations for such an instrument or set of instruments. This need has arisen because of a number of recent developments in broadcasting.

2 Application specific loudness meters

Loudness meters are needed for several applications, including:

- For emission, as a quality-control check.
- To confirm that loudness is within acceptable limits.
- For production, as a level meter to replace the VU meter.

In quality control, a slow (many seconds) averaging meter is needed so that the average programme loudness can be read easily. A numerical readout is an advantage in these applications as it removes any need to interpret needle swings and perform any visual average estimation. A numerical readout also allows corrective gain to be easily determined.

In the production role, particularly if the loudness of complete programme mix is displayed by means of a single meter readout, it may be useful to have some indication of the perceived dynamic range as well as having an indicator fast enough to display individual loudnesses where these can be discriminated aurally. This gives rapid feedback to the programme maker on which to make decisions about individual levels in the mix or overall mix level. This can be particularly important in productions “on location”, where for various reasons, audio monitoring may not be optimal.

For these reasons, a moving-indicator with a short (less than one second) averaging time is needed in production for loudness monitoring. Such a meter could show instantaneous loudness or a slowed average of instantaneous loudness to simplify reading the instrument.

3 Human loudness perception

While loudness models for arbitrary sound types and for arbitrary sound levels are very complicated, ITU-R work has shown that for typical broadcast material and a limited range of loudness, a simplified model can give surprisingly good correlation with perceived average loudness over a 10-15 s interval. There is, so far, no evidence to show that this model cannot also indicate instantaneous loudness under the same constraints. In fact, it is essential to use the same model for instantaneous loudness assessment if results are to be obtained which are consistent with long-term average loudness measurements. The model is detailed in Recommendation ITU-R BS.1770.

The main concern in this Report is adapting this model to indicate instantaneous or short-term loudness.

Human sensory perception follows Stevens' Power Law [Stevens, 1957]. The general form of the law is:

$$\psi(I) = kI^a$$

where:

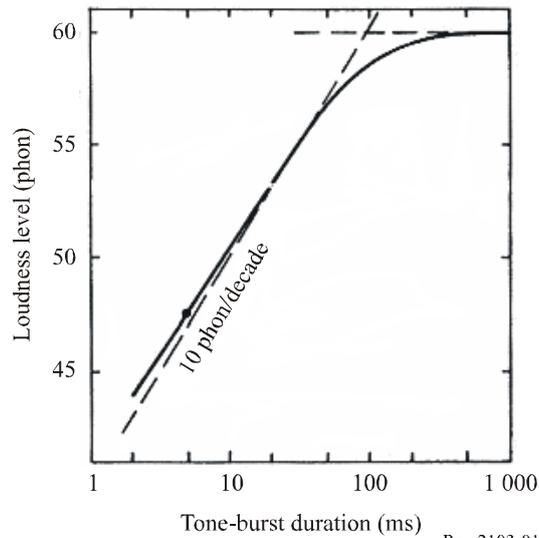
- I*: magnitude of the physical stimulus
- ψ : psychophysical function capturing sensation (the subjective size of the stimulus)
- a*: exponent that depends on the type of stimulation
- k*: proportionality constant that depends on the type of stimulation and the units used.

This law is a consequence of the fact that biological processes, and perception in particular, can usually be modelled by first-order differential equations. These models generally apply to both the magnitude of sensation and to the onset and disappearance of the sensation. In other words, they generally apply in both amplitude and time.

In terms of temporal models of sensation, this means that sensations do not suddenly appear and disappear, but rather the sensation becomes gradually apparent when the stimulus is presented and the sensation gradually disappears after the stimulus is removed. This is illustrated by Zwicker's data on temporal loudness effects for a 2 kHz toneburst stimulus (see Fig. 1) which indicate the relative loudness of a toneburst as a function of its duration. This can also be interpreted as the onset of loudness perception at the start of a continuous tone.

FIGURE 1

Zwicker's data on temporal loudness sensation



Rap 2103-01

4 Review of proposed ballistic types

First order differential equations have solutions for impulse and step stimuli which are simple exponential functions.

Modelling temporal loudness perception can therefore be done with first-order analogue filters which have simple exponential rise and fall characteristics for such stimuli.

This type of filtering can be replicated in the digital domain using a first-order IIR topology or an FIR topology, although the IIR topology is far more computationally efficient. The FIR equivalent has an exponentially falling set of sample weights, corresponding to the impulse response of the first order analogue filter.

It has been suggested in discussions on loudness metering that the long-term loudness algorithm, which uses FIR filtering with rectangular weighting, could be adapted to model instantaneous loudness perception by simply shortening the time window over which it is applied. This would provide a poor model of loudness perception, as illustrated in Figs. 2 and 3.

Figure 2 shows the leading-edge step response of an FIR filter with rectangular weighting and a time window of 100 ms (the best fit for Zwicker's data), compared with a first-order IIR filter with the same time constant. The IIR filter performance follows Zwicker's data within 0.1 dB, while the rectangular FIR filter has a worst-case 2 dB error at 0.1 s. The agreement between the two is otherwise good in the early and late phases.

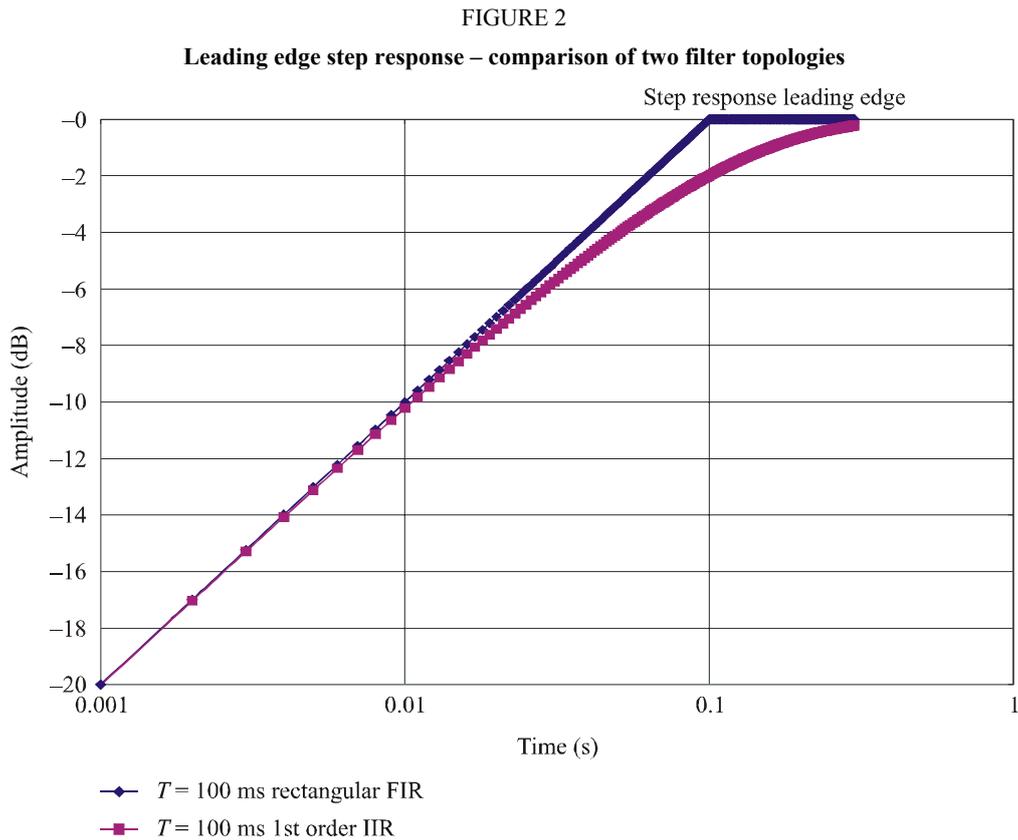
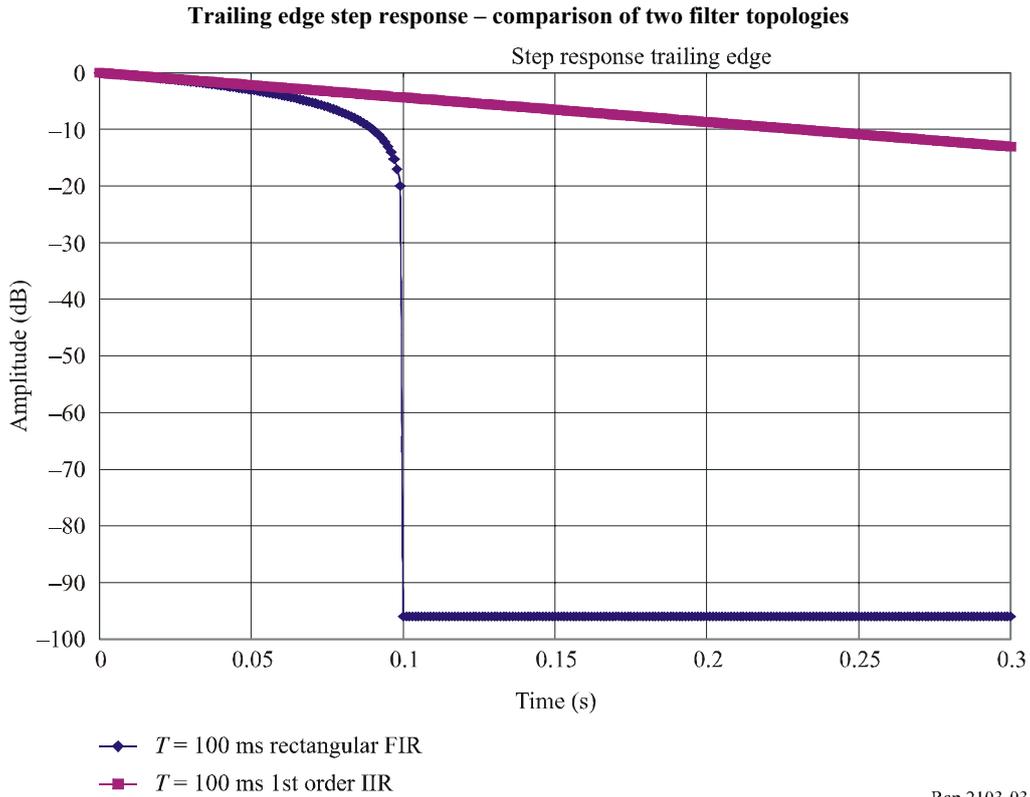


Figure 3 shows a similar comparison for a step response at the trailing edge. Here the disagreement is more marked. While the IIR filter falls off gradually and continuously, the rectangular FIR filter falls gradually at first and then drops precipitously at $T = 100$ ms. This FIR filter is unlikely to be a good model of perception. It gives a very discontinuous representation which does not occur in actual sensation. In other words, it is “jumpy” or “jerky”.

Since human perception generally follows Stevens’ Power Law, and this law represents the behaviour of biological systems which can generally be modelled by first-order differential equations, and since first-order IIR filters model such equations well, and since rectangular-weighted FIR filters do not model such equations well, we should use a first-order IIR filter in any loudness meter that attempts to model instantaneous loudness perception.

FIGURE 3

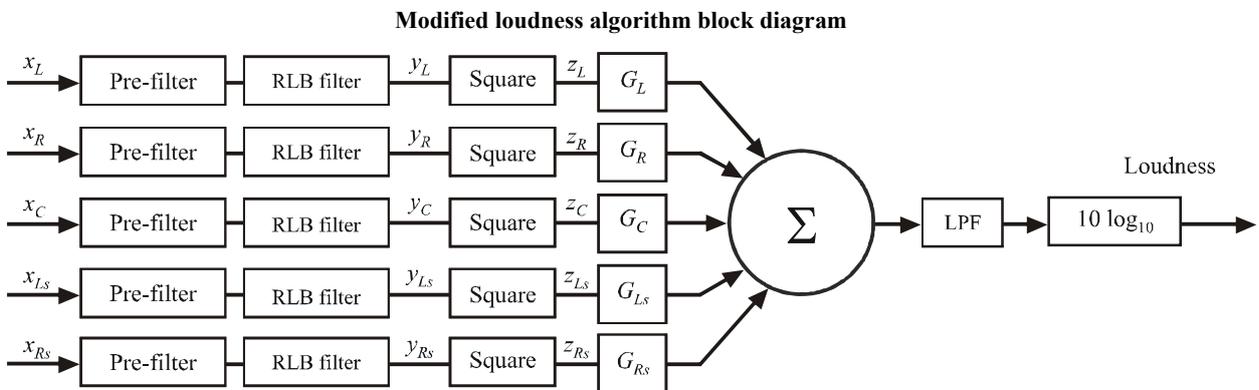


5 Implementation – theory

5.1 Algorithm topology

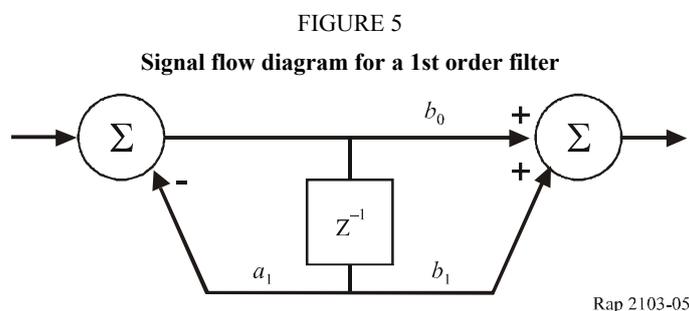
For consistency with the long-term loudness meter, the implementation of the short-term loudness meter followed the block diagram in Fig. 4.

FIGURE 4



5.2 Output filter topology

The low-pass filter stage at the output was functionally equivalent to the first order IIR topology shown in Fig. 5.



5.3 Time constant for instantaneous and fast response

The time constant of the filter should be a minimum of $T = 100$ ms to avoid exaggeration in the modelling of perceived instantaneous loudness. $T = 100$ ms will indicate the perceived dynamic range of the programme. A longer time constant could be used, at the discretion of the operator, to indicate the average loudness with reduced meter swing.

Operational trials in Japan have shown a longer time constant to be preferred over the 100 ms time constant for short-term loudness measurement and indication. It is the view of Japanese experts that a ballistic that changes faster than the commonly used VU meter is not desirable for short-term loudness measurement and indication.

5.4 Audio block decimation

Decimation by averaging blocks of samples may be used to reduce the computation load on the output filter. The decimated sample rate is arbitrary but should be significantly higher than 10 sample/s. Tables 1 and 2 give examples of filter coefficients for a decimated sample rate of 320 samples/s.

TABLE 1

Filter coefficients for first order low-pass filter, $T = 100$ ms

Sample rate	320/s	Gain	6.465674116e + 01
–	–	b_0	1
a_1	-0.9690674172	b_1	1

TABLE 2

Filter coefficients for first order low-pass filter, $T = 400$ ms

Sample rate	320/s	Gain	2.556465999e + 02
–	–	b_0	1
a_1	-0.9921767002	b_1	1

6 Implementation – Loudness meter comparison utility

6.1 Introduction

A software application called Loudness Meter Comparison Utility (LMCU V1.5) has been written by the Australian Broadcasting Corporation to allow comparison of VU, IEC Type II PPM and ITU-R loudness meter characteristics (Recommendation ITU-R BS.1770 – Algorithms to measure audio programme loudness and true-peak audio level). The software runs on a personal computer with the Microsoft Windows XP operating system and may be obtained from the downloads section of the SRG-3 Yahoo egroup (<http://tech.groups.yahoo.com/group/srg3list/>). The LMCU software shows a waveform display (see Fig. 6) for a selected .wav file with an overlaid envelope for each of the three meters, plus a vertical bar graph display of the three meters side by side. Below each bar graph display is a statistical summary of the meter amplitude for the duration of the .wav file, showing median, 75%, 95% and 100% levels. For the loudness meter, arithmetic mean¹ and geometric mean value are also provided. The software allows a first-order low-pass output filter to be selected from a choice of no filter, fast filter ($T = 100$ ms) and slow filter ($T = 400$ ms). The raw loudness data can also be exported to an Excel spreadsheet for further analysis.

The $T = 100$ ms filter (Fast) gives an indication of subjective dynamic range as it corresponds to the temporal integration characteristic of the ear at 2 kHz. This can be quite fatiguing to watch for long periods however, so a second time constant of 400 ms is used to slow down the meter and reduce the size of the swings around the average value. As well as the ergonomic advantages of this approach, it is consistent with sound level metering practice in acoustics.

The software has a play control that allows the .wav file to be played through the standard audio output device. A cursor scrolls across the screen as the item is played.

6.2 Calibration

The levels are calibrated in dB, with the VU meter calibrated so that a sine wave peaking at clipping level reads -3 dB. The loudness meter is also calibrated to read -3 dB for a sine wave peaking at clipping level at 1 kHz. The meter time constants are shown in Table 3.

6.3 Decimation

Audio sample decimation (sample rate reduction) was incorporated in the LMCU software. The decimation algorithm in the LMCU uses a simple arithmetic average for each block of samples. A $\sin(x)/x$ weighting window could be used instead if desired. The block size was set to be significantly less than 100 ms, the time constant of the fast output filter. The block size chosen, 3.125 ms (150 samples at 48 kHz) was arbitrary and shorter blocks could also be used without interfering with the output filter characteristic.

Decimation was incorporated for three reasons:

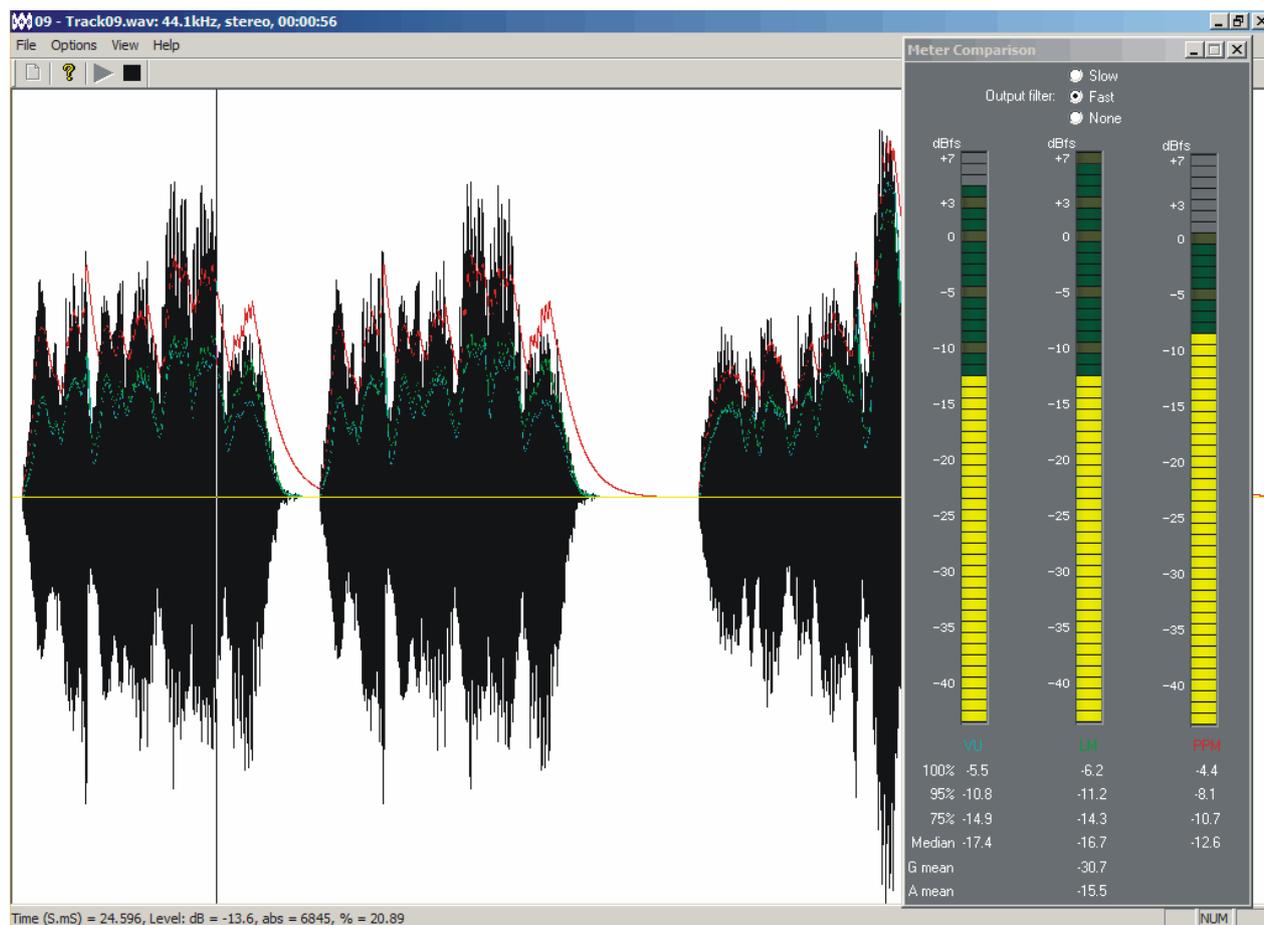
- to reduce the computation load on the output filter;
- to simplify graphic display of the meter envelope;
- to simplify animation of the bar graph display.

The effect of decimation on the mean loudness value is negligible. As it may be a useful tool for loudness meter builders, it is suggested that it should be an optional part of the instantaneous loudness meter specification.

¹ The arithmetic mean loudness is the measure specified in Recommendation.

FIGURE 6

Screen shot of LMCU software (showing track 9 from EBU SQAM test CD)



Rap 2103-06

TABLE 3

Meter time constants

Meter	LP filter order	Rise time	Fall time
PPM (IEC Type II)	1	-2 ± 0.5 dB from peak for 10 ms toneburst at 5 kHz	2.8 ± 0.3 s for 24 dB fall
VU	2	99% of final value in 300 ms, overshoot 1%-1.5%	Symmetrical
Loudness, no filter	–	Instantaneous	Instantaneous
Loudness, fast filter	1	First order low-pass filter, T = 100 ms	Symmetrical
Loudness, slow filter	1	First order low-pass filter, T = 400 ms	Symmetrical

6.4 Gating

For instantaneous loudness indication, gating should not be used. Gating will give a false indication of background noise on wide-range meters during silences.

7 Conclusions

Recent developments in broadcasting have created a need for a new type of level meter which can indicate perceived loudness more accurately than existing level meters. Existing metering systems were developed to indicate either peak level or a combination of peak level and an approximation of perceived loudness. The importance of peak level however has diminished with the introduction of digital audio production and emission methods with their expanded headroom and immunity to over-modulation.

While it is sufficient to know a single-number long-term average loudness for quality control in emission, it is also necessary to have short-term loudness information in production work. This indicates that several types of loudness meter may be needed.

It is important however that these various types of loudness meter give consistent readings. They therefore should use the same loudness measurement algorithm, as specified in Recommendation ITU-R BS.1770. The difference between the types of meter is then reduced to the display format and the temporal characteristic, or ballistic.

While loudness perception is a complex phenomenon, for the purposes of broadcasters it has been found that a relatively simple model, described in Recommendation ITU-R BS.1770, can indicate perceived loudness with surprisingly good accuracy for typical programme material over a limited loudness range. The model conforms with Stevens' Power Law [Stevens, 1957].

Stevens' Power Law is a consequence of the fact that biological systems, and perception in particular, can generally be modelled by first-order differential equations. These apply in the time domain as well as in the amplitude domain.

In the time domain, first-order differential equations can be precisely modelled by first-order analogue filters. Such filters are therefore good models for temporal loudness perception.

A first-order analogue filter can be replicated in the digital domain by a first-order IIR filter or by an FIR filter of arbitrary order with exponentially-decreasing filter weights. The IIR filter is far more computationally efficient however and is therefore preferred.

It has been suggested in discussions on loudness metering that a shortened version of the rectangular-weighted FIR filter used for the long-term loudness algorithm could be used for instantaneous loudness indication. The step response for this filter has been compared with the step response of the first-order IIR filter, both optimized to model Zwicker's data on temporal loudness integration. The IIR filter and the rectangular FIR filter have similar attack characteristics, although with less error in the IIR case. The decay characteristics however differ greatly. The IIR filter has smooth decay. The rectangular FIR filter has an uneven decay, initially smooth, then falling away precipitously. The IIR filter is greatly preferable in this regard for short time constants (100-400 ms). For a very slow indication, an integration window may be a suitable solution.

Implementation details have been given for the IIR filter, using a decimated sample rate. The decimation has no significant effect on the reading as long as the decimated sample rate is well above 10 samples/s.

The IIR filter has been implemented in demonstration software called Loudness Meter Comparison Utility V1.5. The software is described briefly. Readers are encouraged to download and try it for themselves to verify the perceptual accuracy of the filter.

Part 2

1 Purpose

The purpose of this second part is:

- to compare the efficacy of VU and PPM meters with several variants of the algorithm in Recommendation ITU-R BS.1770 – Algorithm to measure audio programme loudness and true-peak audio level, for short-term loudness metering use; and
- to test the effect of differing rise and fall times on loudness metering accuracy to determine a suitable ballistic for short-term loudness metering.

2 Method

The Australian Broadcasting Corporation (ABC) has conducted research into this subject using a software tool called Loudness Meter Comparison Utility (LMCU). This software accepts a sound file in uncompressed wave format, displays the sound file envelope and overlays meter envelopes on this (see Fig. 7). It can also play the sound file through the standard sound output and display three simultaneous bar graphs, one each for VU, PPM and loudness that indicate meter levels at the cursor point as it moves along the sound file. In addition, it calculates statistical information on each meter envelope and can export this information to an Excel spreadsheet for further analysis. This statistical analysis was the basis for the correlation figures in the following section.

An update of this tool has been produced (V1.6) which includes the following features:

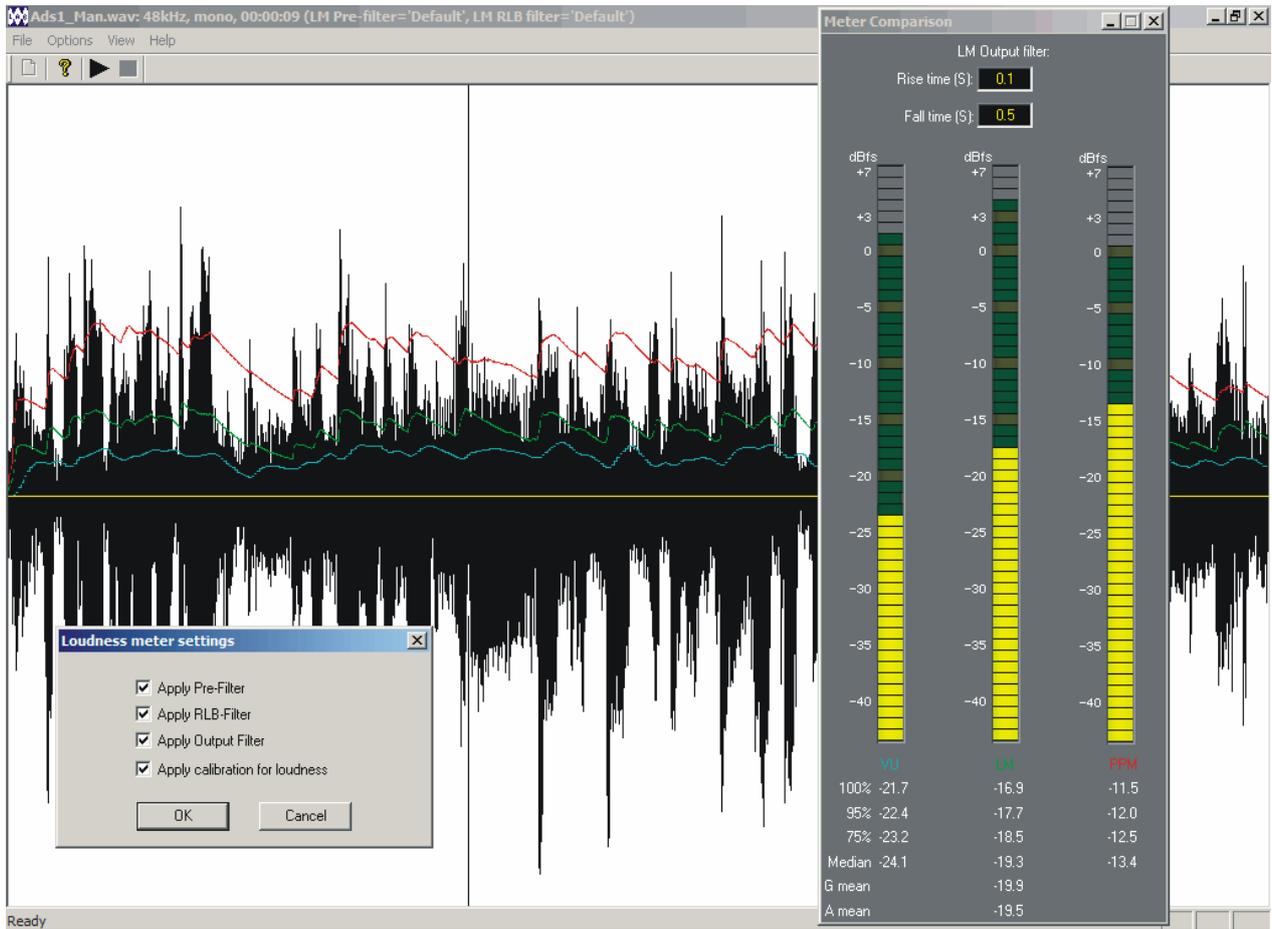
- separate user-settable rise and fall times on the loudness meter;
- enable-disable options on pre-filtering, RLB filtering, calibration constant and output filtering;
- user-settable filter coefficients for the pre-filtering and RLB filtering functions.

The output filter characteristic is a first-order IIR digital type on both rise and fall, rather than the arithmetic average method specified in Recommendation ITU-R BS.1770 for long-term loudness measurement. As discussed in Part 1, the first-order filter type ensures a smooth fall characteristic and closer conformance with established temporal loudness perception data for tonebursts.

A number of weighting filter types and rise/fall time constants have been tested with this software. The results have been compared with the results from subjective listening tests held in 2003*. The comparison uses the familiar statistical correlation measure, as the 2003 report showed that it gave the same ranking as virtually every other statistical measure used. The comparison used the 49 monophonic ITU test samples from the 2003 subjective test. The full subjective test results were not available, so a subset of the results was used, taken from 41 listeners at the ABC and at the Australian National Acoustic Laboratories. While the variances in this subset were higher than for the overall result set, the means were in fairly close agreement and the correlation figures for previously-tested algorithm variants are similar. The reference item for these tests was the sound file “Female Interviewer.wav”.

* ITU-R WP 6P SRG-3 Status Report 5, Document 6P/15, 16 September 2003.

FIGURE 7
loudness meter comparison utility V1.6 screen



Rap 2103-07

3 Results

3.1 Meter types

Five meter types were tested:

- 1 Volume indicator or “VU”.
- 2 Peak programme meter or “PPM”.
- 3 Unweighted RMS loudness.
- 4 BS.1770-weighted loudness.
- 5 ABC/USyd-weighted loudness.

The last type was an RMS algorithm using a weighting curve derived from a 5 channel listening test using octave-band noise, conducted jointly by the ABC and the University of Sydney [Dash *et al.*, 2008]. The weighting curve for this filter is shown in Fig. 8. All three loudness meters were based on the BS.1770 algorithm, varying only in their frequency weighting.

3.2 VU and PPM results

The VU and PPM simulations were read by three methods based on the 50%, 75% and 95% points on the cumulative distributions of the sampled meter readings. The 50% measure is the same as a median. The operational equivalent of this method of reading is time spent below a given meter level, so 75% represents the point on the meter scale where the indicator spent 75% of the time below and 25% of the time above. The 50% point represents approximately the midpoint of the indicator swings, while the 95% point represents near the top of the swings. The correlations of these results with perceived programme loudness are plotted in Fig. 9.

Correlation is a dimensionless number however, and it does not give information on the absolute accuracy of a model, only the relative accuracy, so for the same results, the absolute RMS errors (dB) are plotted in the same figure using the average of all item levels as a reference. These results vary slightly from the correlation results as the latter come from a line of best fit, which may not have unity slope, while the RMS error in this case is derived from a line of unity slope. The trends however are fairly consistent between the two methods.

Figure 10 shows the effect of the reading method on magnitude of the average reading for the VU and the PPM. For the PPM, the steps between reading methods are around 1 dB. For the VU the steps are about 2 and 5 dB.

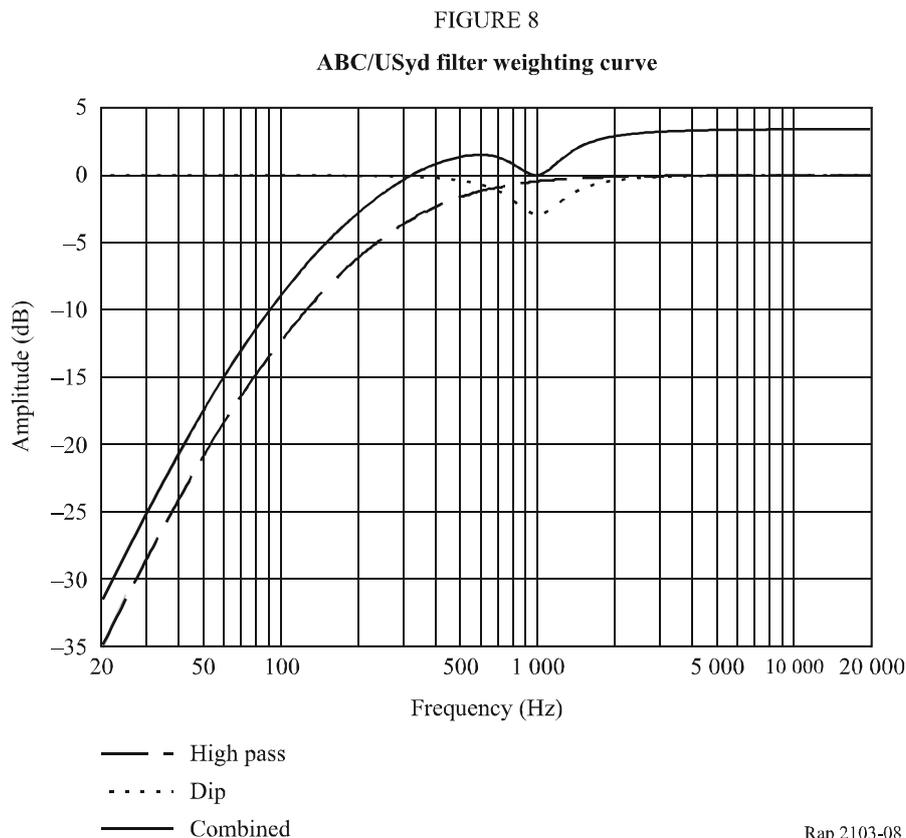


FIGURE 9

VU and PPM correlations with perceived programme loudness for three reading methods

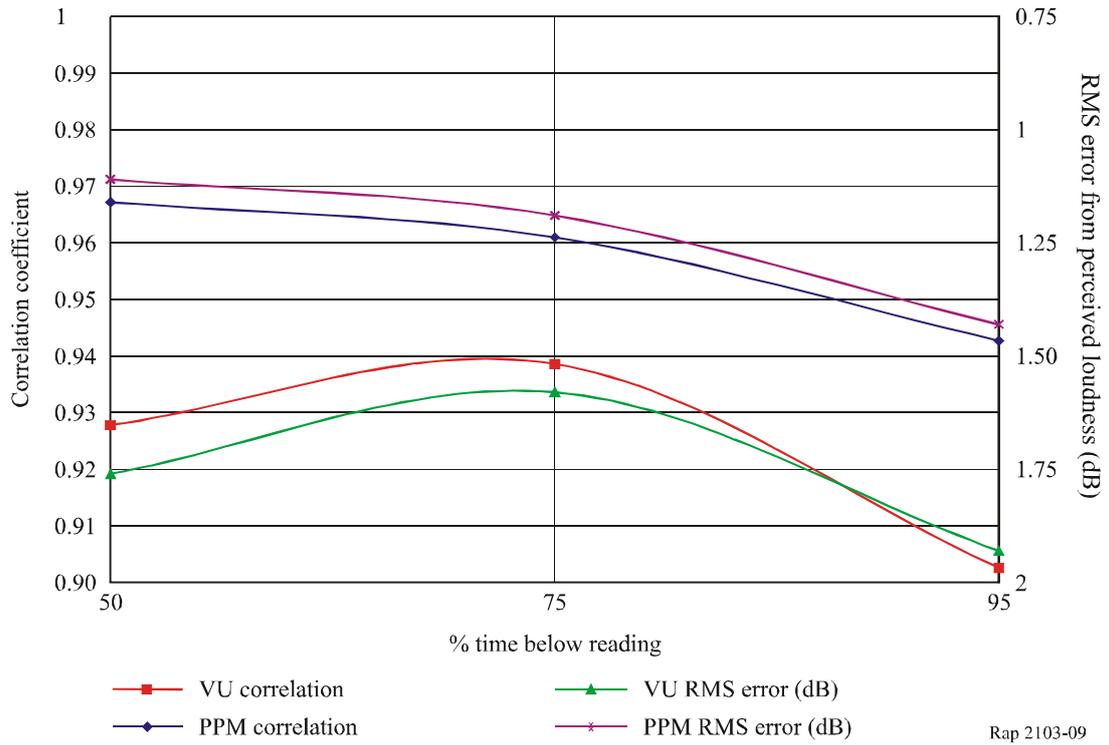
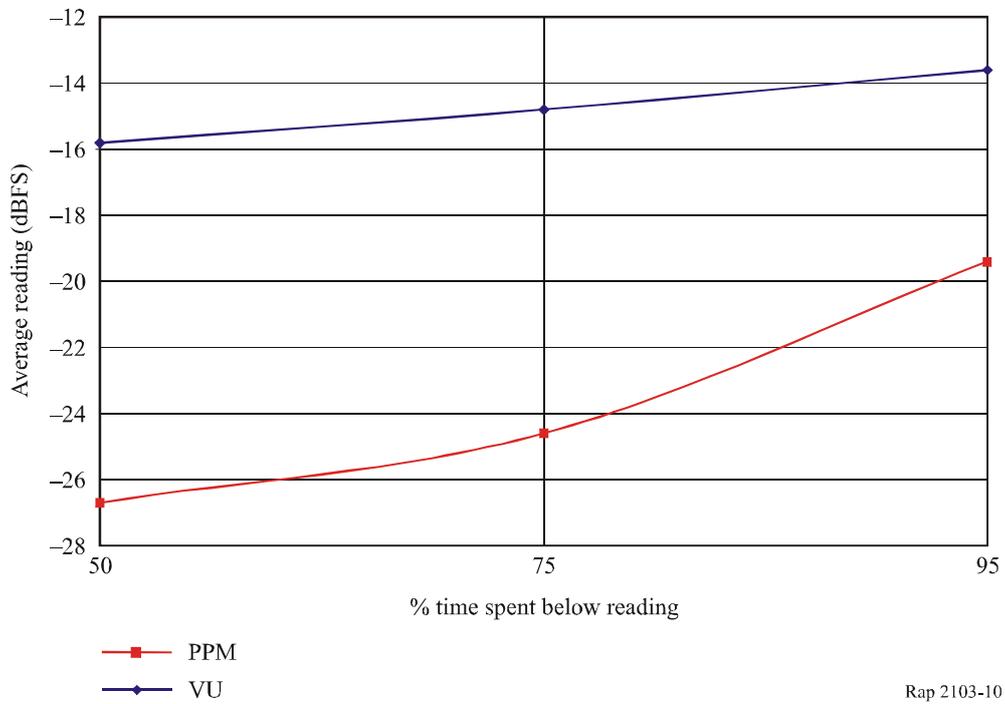


FIGURE 10

Average reading vs. reading method



3.3 Loudness meter results

Figure 11 shows a comparison of the effect of three different weighting filters on the loudness reading of a Recommendation ITU-R BS.1770-type meter, without any ballistic effect. This is shown both as a correlation with perceived programme loudness and as an RMS error in dB. Performance of the Recommendation ITU-R BS.1770 *K*-weighting filter is similar to an unweighted measurement using both metrics, and both are significantly better than the ABC/USyd filter in this context.

Next, the effect of varying ballistic characteristic was tested in several variants of loudness meter to determine agreement with subjective perception of long-term programme loudness i.e. overall loudness for the entire programme segment. While this is no guarantee of accuracy in representing perception of short-term loudness, it does at least allow consistency of the two forms of metering to be compared. Since the primary aim of specifying a short-term loudness meter is to have a compatible production tool to use for control of long-term loudness, accurate representation of short-term loudness, while desirable, is not the most important characteristic of a short-term loudness meter in a broadcasting context.

Because an RMS summation requires averaging before power summation, this filtering should be performed separately for each channel before summing partial loudnesses, as shown in Fig. 12. In the test results here, only mono testing was considered, so the results here are not affected by this issue.

FIGURE 11

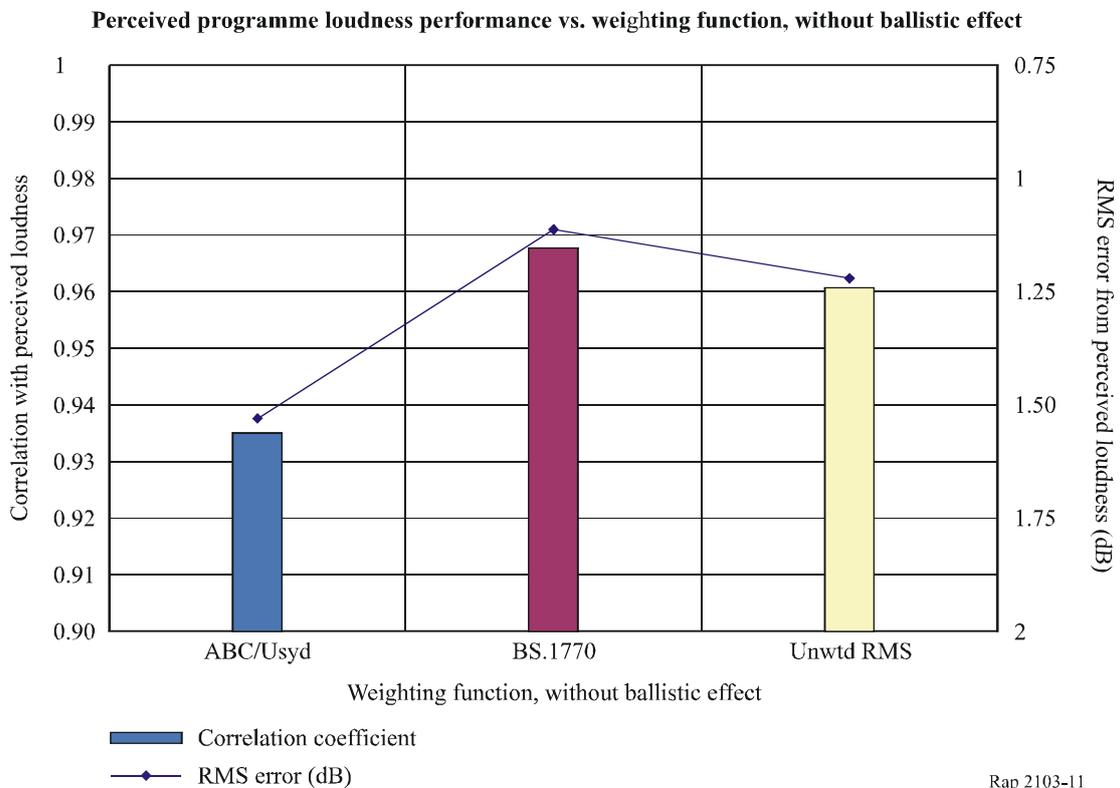
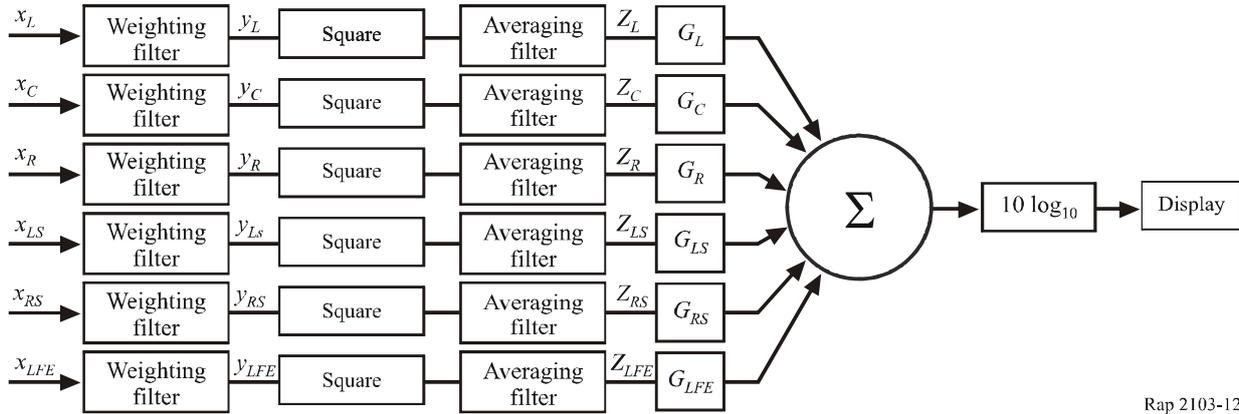


FIGURE 12

Adaptation of the BS.1770 loudness algorithm topology for short-term loudness assessment

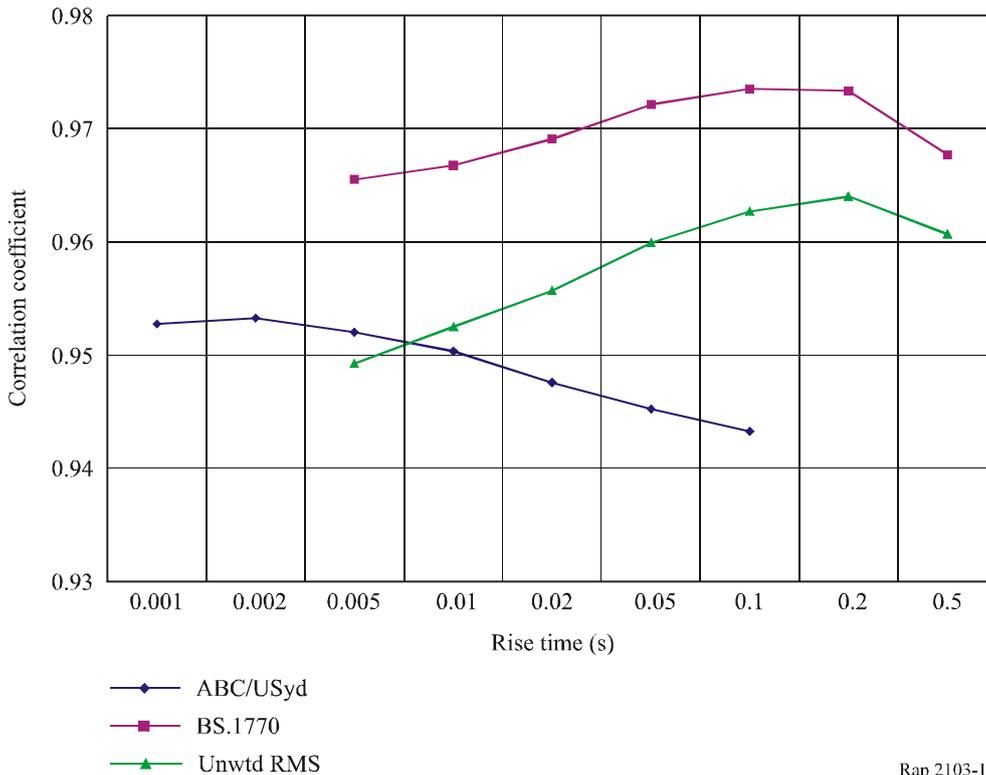


Rap 2103-12

Figures 13 and 14 show the effects of varying rise and fall ballistic times on the performance of these filters. As discussed in Part 1, the output filter characteristic for a short-term loudness meter should be first-order analogue or first-order IIR digital rather than the simple average method specified in Recommendation ITU-R BS.1770 for long-term loudness measurement. The first-order filter type ensures a smooth fall characteristic and closer conformance with established temporal loudness perception data for tonebursts.

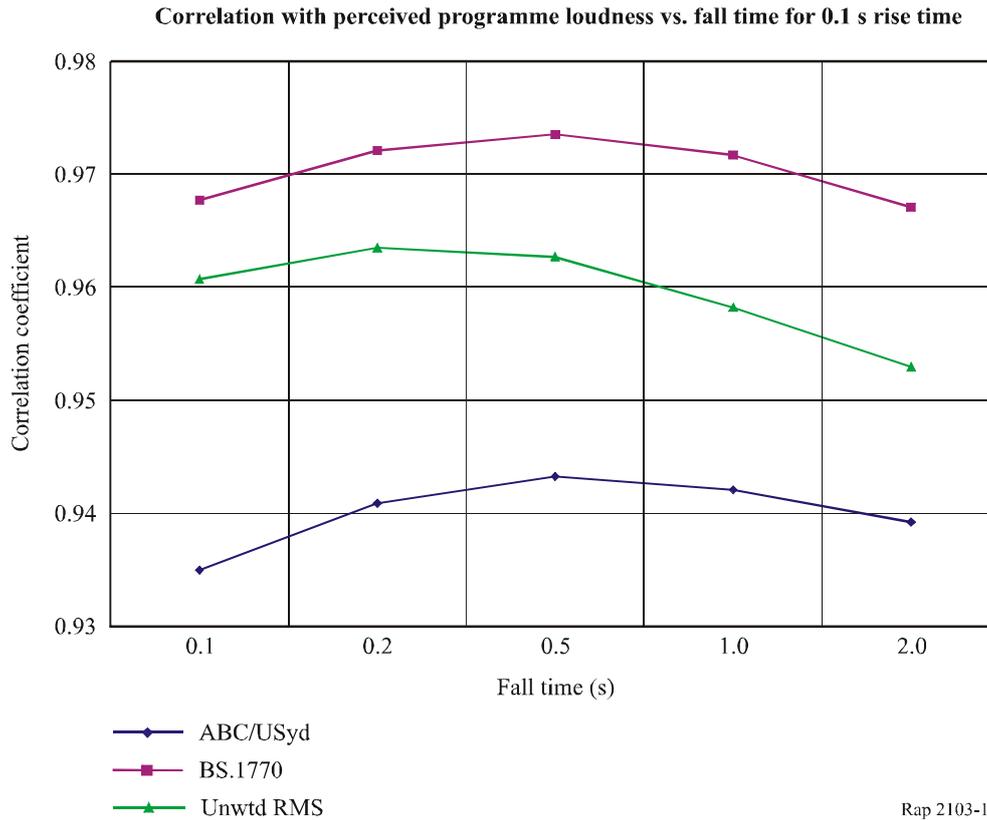
FIGURE 13

Correlation with perceived programme loudness vs. rise time for 0.5 s fall time



Rap 2103-13

FIGURE 14



The ballistics are therefore based on first-order filters, and the time constants are exponential decay times. These filters are implemented after the signal squaring rectifier function, so their rise and fall characteristics are equivalent to filters of order 1/2 in the linear signal domain.

The Recommendation ITU-R BS.1770 filter and the unweighted RMS measure both show best agreement with perceived programme loudness at rise times of around 0.1-0.2 s while the ABC/USyd filter shows best agreement at very short rise times around 2 ms. All three filters show best agreement with perceived programme loudness at fall times of around 0.2-0.5 s. The effect of rise and fall time on this agreement however is weak in all cases – much weaker than the effect of the frequency weighting.

While the ABC/USyd meter weighting did not correlate as well with long-term loudness judgments as the Recommendation ITU-R BS.1770 weighting did, the former appeared to better reflect the instantaneous loudness when watching the bar graph indicator.

Numerical results for these filters and time constants are shown in Tables 4 to 10. The reference item levels indicate the relative gains of the algorithms.

Two time constant combinations are of particular interest from prior research. Bauer *et al* [1967] of CBS Laboratories Bauer and Torick [1966] concluded that a rise time of 0.1 s and a fall time of 0.5 s was optimal for short-term loudness metering. Corresponding figures in Tables 6 to 8 are shown in red. Glasberg and Moore [2002] later produced an even more sophisticated loudness model which included rise time of 0.1 s and fall time of 2.0 s for long-term loudness assessment². Corresponding figures in Tables 6 to 8 are shown in blue. Optimal performance for each filter type is indicated by bold figures.

² Glasberg and Moore derived their long-term loudness measure from an average of a short-term loudness measure, which is the opposite of the approach adopted in ITU work on this topic.

The correlation figures for these time constants should not be taken as a measure of validity of prior research, as the loudness models and evaluation methods used in those works were quite different from the present models. The comparison is considered mainly for the purpose of examining consistency of the present results with existing systems. When looking purely at time constants, the results for Recommendation ITU-R BS.1770 weighting and unweighted RMS algorithms show good consistency with the CBS model and slightly less consistency with the Glasberg and Moore model. The time-constant results for the ABC/USyd weighting show less consistency with the CBS and Glasberg and Moore models, even though the frequency weighting in the ABC/USyd model is closer to the frequency weightings used in those models.

Finally, Figs. 15 and 16 shows the average reading for all programme items as a function of rise and fall time. Figure 15 shows average reading vs. rise time for a fixed fall time of 0.5 s. Figure 16 shows average reading vs. fall time for a fixed rise time of 0.1 s. The curves for different weighting functions are all consistent and indicate the effect of asymmetry in the ballistic time constants. If an asymmetrical time constant were chosen, a calibration factor would need to be incorporated to maintain consistency with the long-term loudness averaging method.

FIGURE 15

Average level vs. rise time for 0.5 s fall time

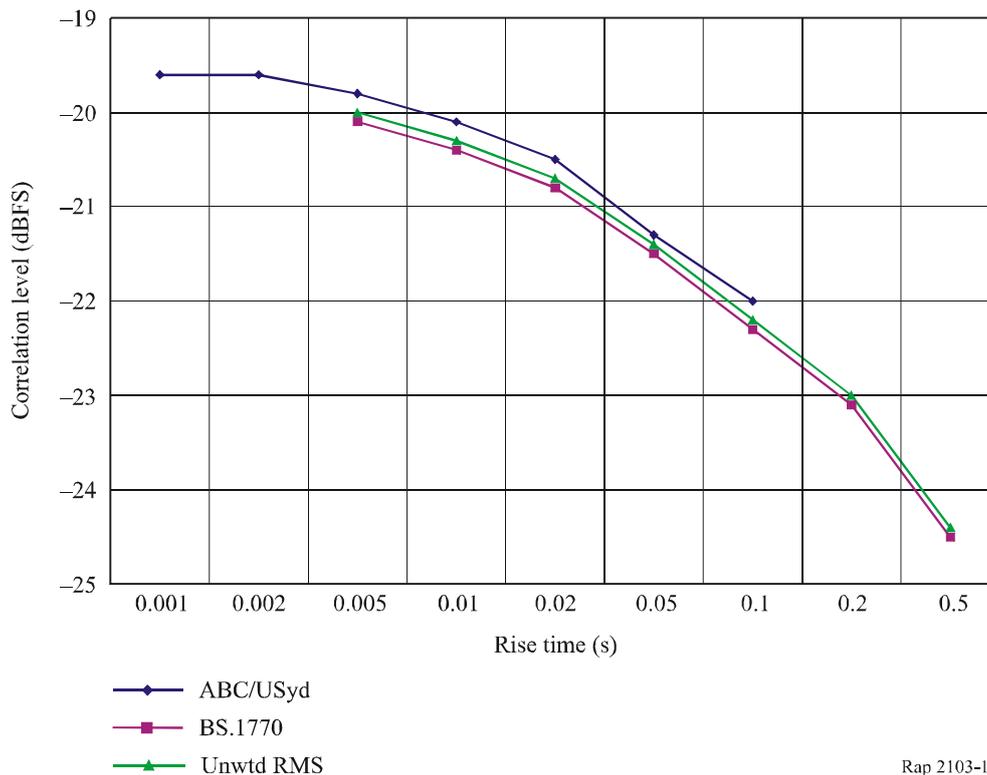
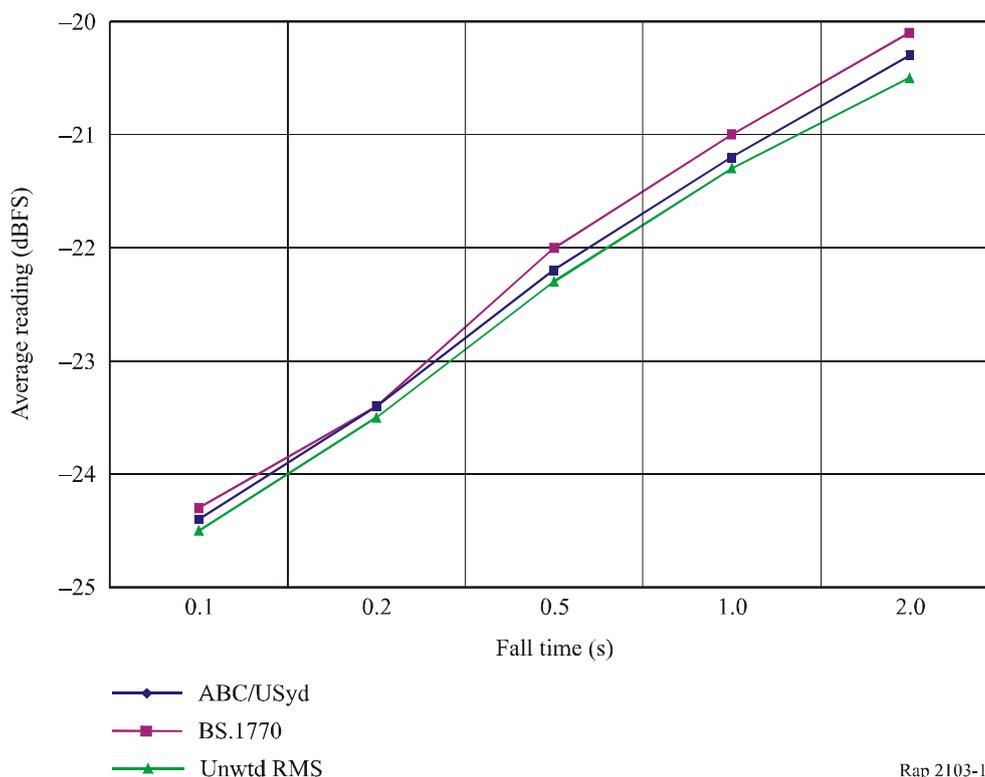


FIGURE 16

Average level vs. fall time for 0.1 s rise time



4 Conclusions

- a) As discussed in Report ITU-R BS.2054 – Audio levels and loudness, the output filter characteristic for a short-term loudness meter should be first-order analogue or first-order IIR digital rather than the arithmetic average method specified in Recommendation ITU-R BS.1770 for long-term loudness measurement. The first-order filter type ensures a smooth fall characteristic and closer conformance with established temporal loudness perception data for tonebursts.
- b) Because an RMS summation requires averaging before power summation, this filtering should be performed separately for each channel before summing partial loudnesses, as shown in Fig. 6.
- c) When comparing VU and PPM meters and loudness algorithms without ballistic effects, the 50% PPM and the Recommendation ITU-R BS.1770 algorithm are equally accurate at assessing loudness and the unweighted RMS algorithm is almost as effective. In broadcasting facilities already equipped with PPMs, it is recommended that this form of metering should be used where possible for short-term loudness assessment on individual channels. In broadcasting facilities equipped with VU meters, these meters may be used for assessing loudness although allowance should be made for their larger difference from subjective loudness judgments.

In both cases, allowance should be made for the magnitude difference between the readings of these meters and the readings of loudness meters. In the case of the 50% PPM reading, the reading is around 8 dB higher than a loudness meter. In the case of the 75% VU reading, the average reading is around the same as for a loudness meter.

The operator is reminded that a loudness meter is read at the midpoint of its swing i.e. the average reading. This also applies to PPMs when used for loudness assessment³. This differs from the convention used for reading a VU meter and for reading peaks from a PPM.

- d) Since PPMs are not designed for combined multi-channel signal indication, overall short-term loudness indication on a multi-channel system should be provided by a multi-channel short-term version of an RMS-indicating loudness meter either with Recommendation ITU-R BS.1770 weighting or with no weighting.
- e) Of the time constants tested, a rise time of 0.1 s and a fall time of 0.5 s gave optimal performance with the BS.1770 algorithm and near-optimal performance with the unweighted RMS algorithm. The effect of rise and fall time on the loudness correlation appears to be weak however, and loudness correlation is almost as good with symmetrical rise and fall time constants. These will give readings which are more directly compatible with the long-term version of the Recommendation ITU-R BS.1770 algorithm⁴ and they are therefore recommended. As the average reading of such a meter is largely unaffected by the time constant, this choice is left open to the manufacturer and/or the user, but a minimum of 0.1 s is recommended both to avoid exaggerated indication of perceived dynamic range and to minimize visual fatigue.

5 References and Bibliography

- BAUER B.B. and TORICK E.L. [September 1966] Researches in loudness measurement. *IEEE Trans Audio and Electroacoustics*, Vol. AU-14, **3**, p. 141-151.
- BAUER B.B., TORICK E.L., ROSENHECK A.J. and ALLEN R.G. [December 1967] A loudness-level monitor for broadcasting. *IEEE Trans Audio and Electroacoustics*, Vol. AU-15, **4**, p. 177-182.
- DASH I., MIRANDA L. and CABRERA D. [17-20 May 2008] Multichannel Loudness Listening Test. Audio Engineering Society Convention Paper Presented at the 124th Convention, Amsterdam, The Netherlands.
- GLASBERG B. and MOORE B.C.J. [May 2002] A model of loudness applicable to time-varying sounds. *J. Aud. Eng. Soc.*, Vol. 50, **5**, p. 331-341.
- STEVENS S.S. [1957] On the psychophysical law. *Psychological Rev.* 64, **3** p.153-181. A précis is available at http://en.wikipedia.org/wiki/Stevens%27_power_law.

ITU-R texts

Recommendation ITU-R BS.1771 – Requirements for loudness and true-peak indicating meters.

³ This is consistent with the BBC practice of setting gain by the PPM swing rather than just the peak indication.

⁴ Correlation only shows relative agreement, or errors from the line of best fit – it gives no information on absolute offset. Two algorithms with similar correlation figures can still have very different gains.

Appendix 1

Results in numerical form

TABLE 4

Standard VU and IEC Type II PPM correlations with subjective programme loudness assessment

Model	VU 50%	VU 75%	VU 95%
Correlation	0.9278	0.9386	0.9025
RMS error (dB)	1.76	1.58	1.93
Average item level (dBFS)	-26.7	-24.6	-19.4
Model	PPM 50%	PPM 75%	PPM 95%
Correlation	0.9672	0.9610	0.9427
RMS error (dB)	1.11	1.19	1.43
Average item level (dBFS)	-15.8	-14.8	-13.6

TABLE 5

Correlations with perceived programme loudness weighting functions without ballistic effect

Model	Unwtd RMS	BS.1770	ABC/USyd wtg
Correlation	0.9607	0.9677	0.9350
RMS error (dB)	1.22	1.11	1.53
Average item level (dBFS)	-24.4	-24.5	-24.6

TABLE 6

Correlations with perceived programme loudness for rise and fall time combinations – unweighted RMS

Fall (s)	0.1	0.2	0.5	1.0	2.0
Rise (s)					
0.5			0.9607	0.9634	0.9610
0.2		0.9607	0.9640	0.9614	0.9567
0.1	0.9607	0.9635	0.9627	0.9582	0.9529
0.05			0.9599		
0.02			0.9557		
0.01			0.9525		
0.005			0.9493		

TABLE 7

**Correlations with perceived programme loudness for rise
and fall time combinations – BS.1770 weighting**

Fall (s)	0.1	0.2	0.5	1.0	2.0
Rise (s)					
0.5			0.9677	0.9730	0.9731
0.2		0.9677	0.9733	0.9729	0.9702
0.1	0.9677	0.9721	0.9735	0.9717	0.9671
0.05			0.9722		
0.02			0.9691		
0.01			0.9668		
0.005			0.9655		

TABLE 8

**Correlations with perceived programme loudness for rise
and fall time combinations – ABC/USyd weighting**

Fall (s)	0.1	0.2	0.5	1.0	2.0
Rise (s)					
0.1	0.9350	0.9409	0.9433	0.9421	0.9392
0.05			0.9452		
0.02			0.9476		
0.01		0.9527	0.9504		
0.005	0.9534	0.9545	0.9520	0.9483	
0.002	0.9547	0.9559	0.9533		
0.001	0.9551	0.9567	0.9527		

TABLE 9

**Correlations with perceived programme loudness as a function
of rise time – all weightings, fall time 0.5 s**

Rise (s)	Correlation comparison for 0.5 s fall time		
	ABC/Usyd	BS.1770	Unwtd RMS
0.001	0.9527		
0.002	0.9533		
0.005	0.9520	0.9655	0.9493
0.01	0.9504	0.9668	0.9525
0.02	0.9476	0.9691	0.9557
0.05	0.9452	0.9722	0.9599
0.1	0.9433	0.9735	0.9627
0.2		0.9733	0.9640
0.5		0.9677	0.9607

TABLE 10

Average level as a function of rise time – all weightings, fall time 0.5 s

Rise (s)	Average level (dBFS), 0.5 s fall time		
	ABC/Usyd	BS.1770	Unwtd RMS
0.001	-19.6		
0.002	-19.6		
0.005	-19.8	-20.1	-20.0
0.01	-20.1	-20.4	-20.3
0.02	-20.5	-20.8	-20.7
0.05	-21.3	-21.5	-21.4
0.1	-22.0	-22.3	-22.2
0.2		-23.1	-23.0
0.5		-24.5	-24.4

TABLE 11

Average level as a function of fall time – all weightings, rise time 0.1 s

Fall time (s)	ABC/Usyd	BS.1770	Unwtd RMS
S	dBFS	dBFS	dBFS
0.1	-24.3	-24.5	-24.4
0.2	-23.4	-23.5	-23.4
0.5	-22	-22.3	-22.2
1.0	-21	-21.3	-21.2
2.0	-20.1	-20.5	-20.3