



**Recommendation ITU-R BS.1770-4**  
**(10/2015)**

**Algorithms to measure audio programme  
loudness and true-peak audio level**

**BS Series**  
**Broadcasting service (sound)**



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## Foreword

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(Also available online at <http://www.itu.int/publ/R-REC/en>)

| Series     | Title                                                                                |
|------------|--------------------------------------------------------------------------------------|
| <b>BO</b>  | Satellite delivery                                                                   |
| <b>BR</b>  | Recording for production, archival and play-out; film for television                 |
| <b>BS</b>  | <b>Broadcasting service (sound)</b>                                                  |
| <b>BT</b>  | Broadcasting service (television)                                                    |
| <b>F</b>   | Fixed service                                                                        |
| <b>M</b>   | Mobile, radiodetermination, amateur and related satellite services                   |
| <b>P</b>   | Radiowave propagation                                                                |
| <b>RA</b>  | Radio astronomy                                                                      |
| <b>RS</b>  | Remote sensing systems                                                               |
| <b>S</b>   | Fixed-satellite service                                                              |
| <b>SA</b>  | Space applications and meteorology                                                   |
| <b>SF</b>  | Frequency sharing and coordination between fixed-satellite and fixed service systems |
| <b>SM</b>  | Spectrum management                                                                  |
| <b>SNG</b> | Satellite news gathering                                                             |
| <b>TF</b>  | Time signals and frequency standards emissions                                       |
| <b>V</b>   | Vocabulary and related subjects                                                      |

**Note:** This ITU-R Recommendation was approved in English under the procedure detailed in Resolution ITU-R 1.

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## RECOMMENDATION ITU-R BS.1770-4

**Algorithms to measure audio programme  
loudness and true-peak audio level**

(Question ITU-R 2/6)

(2006-2007-2011-2012-2015)

**Scope**

This Recommendation specifies audio measurement algorithms for the purpose of determining subjective programme loudness, and true-peak signal level.

The ITU Radiocommunication Assembly,

*considering*

- a) that modern digital sound transmission techniques offer an extremely wide dynamic range;
- b) that modern digital sound production and transmission techniques provide a mixture of mono, stereo and 3/2 multichannel formats specified in Recommendation ITU-R BS.775 and advanced sound formats specified in Recommendation ITU-R BS.2051, and that sound programmes are produced in all of these formats;
- c) that listeners desire the subjective loudness of audio programmes to be uniform for different sources and programme types;
- d) that many methods are available for measurement of audio levels but that existing measurement methods employed in programme production do not provide indication of subjective loudness;
- e) that, for the purpose of loudness control in programme exchange, in order to reduce audience annoyance, it is essential to have a single recommended algorithm for objective estimation of subjective loudness;
- f) that future complex algorithms based on psychoacoustic models may provide improved objective measures of loudness for a wide variety of audio programmes;
- g) that digital media overload abruptly, and thus even momentary overload should be avoided,

*considering further*

- h) that peak signal levels may increase due to commonly applied processes such as filtering or bit-rate reduction;
- j) that existing metering technologies do not reflect the true-peak level contained in a digital signal since the true-peak value may occur in between samples;
- k) that the state of digital signal processing makes it practical to implement an algorithm that closely estimates the true-peak level of a signal;
- l) that use of a true-peak indicating algorithm will allow accurate indication of the headroom between the peak level of a digital audio signal and the clipping level,

*recommends*

- 1 that when an objective measure of the loudness of an audio channel or programme, produced with up to 5 main channels per Recommendation BS.775 (mono source, stereo and 3/2 multichannel sound), is required to facilitate programme delivery and exchange, the algorithm specified in Annex 1 should be used;
- 2 that when an objective measure of the loudness of an audio programme produced with a larger number of channels (such as the channel configurations specified in Recommendation ITU-R BS.2051) is required, the algorithm specified in Annex 3 should be used;
- 3 that methods employed in programme production and post-production to indicate programme loudness may be based on the algorithm specified in Annex 1 and Annex 3;
- 4 that when an indication of true-peak level of a digital audio signal is required, the measurement method should be based on the guidelines shown in Annex 2, or on a method that gives similar or superior results,

NOTE 1 – Users should be aware that measured loudness is an estimation of subjective loudness and involves some degree of uncertainty depending on listeners, audio material and listening conditions.

*further recommends*

- 1 that consideration should be given to the possible need to update this Recommendation in the event that new loudness algorithms are shown to provide performance that is significantly improved over the algorithm specified in Annex 1 and Annex 3;
- 2 that this Recommendation should be updated when new algorithms have been developed to enable the measurement of audio programme loudness for object and scene based audio programmes.

NOTE 2 – For testing compliance of meters according to this Recommendation, test material from the set described in Report ITU-R BS.2217 may be used.

## Annex 1

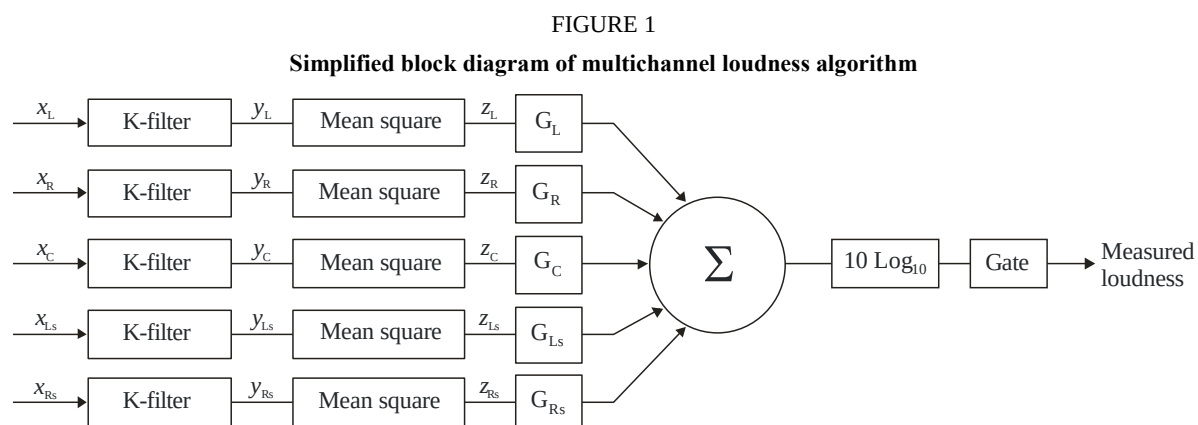
### Specification of the objective multichannel loudness measurement algorithm

This Annex specifies the multichannel loudness measurement modelling algorithm.

The algorithm consists of four stages

- “K” frequency weighting;
- mean square calculation for each channel;
- channel-weighted summation (surround channels have larger weights, and the LFE channel is excluded);
- gating of 400 ms blocks (overlapping by 75%), where two thresholds are used:
  - the first at –70 LKFS;
  - the second at –10 dB relative to the level measured after application of the first threshold.

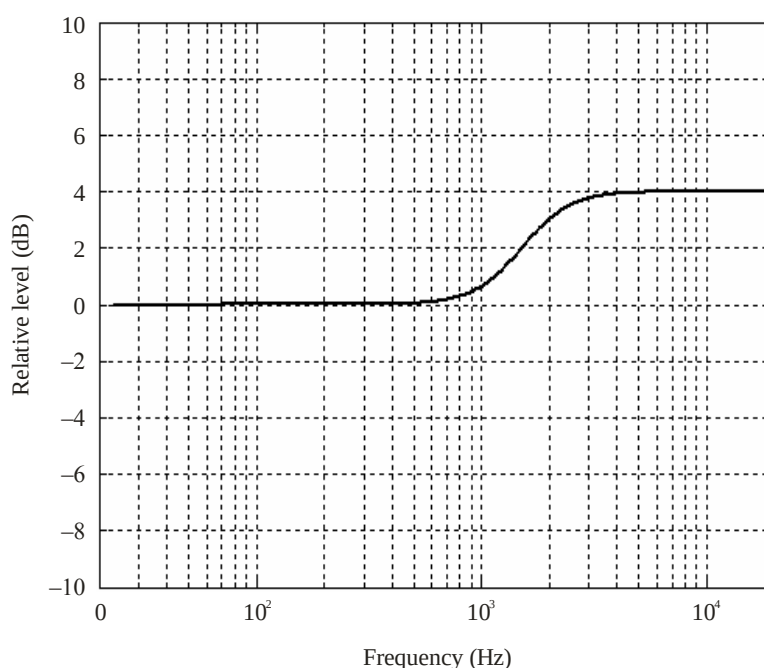
Figure 1 shows a block diagram of the various components of the algorithm. Labels are provided at different points along the signal flow path to aid in the description of the algorithm. The block diagram shows inputs for five main channels (left, centre, right, left surround and right surround); this allows monitoring of programmes containing from one to five channels. For a programme that has less than five channels some inputs would not be used. The low frequency effects (LFE) channel is not included in the measurement.



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The first step of the algorithm applies a 2-stage pre-filtering<sup>1</sup> of the signal. The first stage of the pre-filtering accounts for the acoustic effects of the head, where the head is modelled as a rigid sphere. The response is shown in Fig. 2.

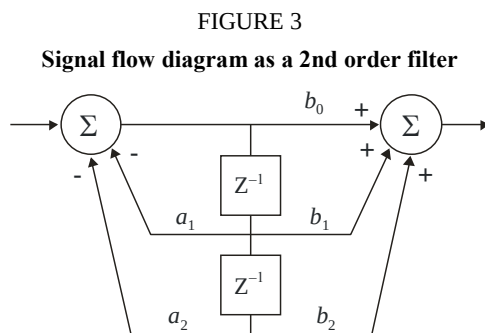
FIGURE 2  
Response of stage 1 of the pre-filter used to account for the acoustic effects of the head



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<sup>1</sup> The K-weighting filter is composed of two stages of filtering; a first stage shelving filter and a second stage high-pass filter.

Stage 1 of the pre-filter is defined by the filter shown in Fig. 3 with the coefficients specified in Table 1.



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TABLE 1

**Filter coefficients for stage 1 of the pre-filter to model a spherical head**

|       |                   |       |                   |
|-------|-------------------|-------|-------------------|
|       |                   | $b_0$ | 1.53512485958697  |
| $a_1$ | -1.69065929318241 | $b_1$ | -2.69169618940638 |
| $a_2$ | 0.73248077421585  | $b_2$ | 1.19839281085285  |

These filter coefficients are for a sampling rate of 48 kHz. Implementations at other sampling rates will require different coefficient values, which should be chosen to provide the same frequency response that the specified filter provides at 48 kHz. The values of these coefficients may need to be quantized due to the internal precision of the available hardware. Tests have shown that the performance of the algorithm is not sensitive to small variations in these coefficients.

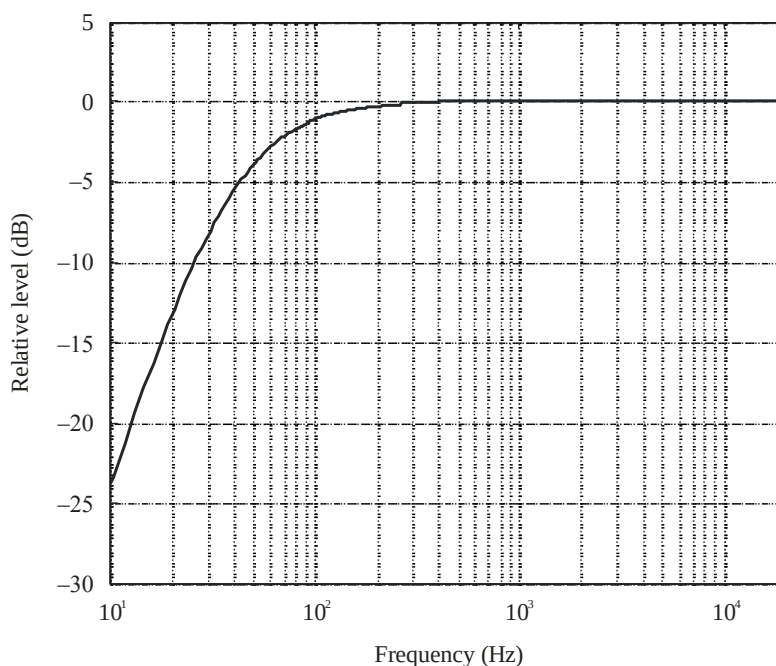
The second stage of the pre-filter applies a simple high-pass filter as shown in Fig. 4.

The stage weighting curve is specified as a 2nd order filter as shown in Fig. 3, with the coefficients specified in Table 2.

These filter coefficients are for a sampling rate of 48 kHz. Implementations at other sampling rates will require different coefficient values, which should be chosen to provide the same frequency response that the specified filter provides at 48 kHz.

FIGURE 4

Second stage weighting curve



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TABLE 2

Filter coefficients for the second stage weighting curve

|       |                   |       |      |
|-------|-------------------|-------|------|
|       |                   | $b_0$ | 1.0  |
| $a_1$ | -1.99004745483398 | $b_1$ | -2.0 |
| $a_2$ | 0.99007225036621  | $b_2$ | 1.0  |

The power, the mean square of the filtered input signal in a measurement interval  $T$  is measured as:

$$z_i = \frac{1}{T} \int_0^T y_i^2 dt \quad (1)$$

where  $y_i$  is the input signal (filtered by the 2-stage pre-filter as described above), and  $i \in I$  where  $I = \{L, R, C, Ls, Rs\}$ , the set of input channels.

The loudness over the measurement interval  $T$  is defined as:

$$\text{Loudness, } L_K = -0.691 + 10 \log_{10} \sum_i G_i \cdot z_i \quad \text{LKFS} \quad (2)$$

where  $G_i$  are the weighting coefficients for the individual channels.

To calculate a gated loudness measurement, the interval  $T$  is divided into a set of overlapping gating block intervals. A gating block is a set of contiguous audio samples of duration  $T_g = 400$  ms, to the nearest sample. The overlap of each gating block shall be 75% of the gating block duration.

The measurement interval shall be constrained such that it ends at the end of a gating block. Incomplete gating blocks at the end of the measurement interval are not used.

The power, the mean square of the  $j$ th gating block of the  $i$ th input channel in the interval  $T$  is:

$$z_{ij} = \frac{1}{T_g} \int_{T_g \cdot j \cdot \text{step}}^{T_g \cdot (j \cdot \text{step} + 1)} y_i^2 dt \quad \text{where} \quad \text{step} = 1 - \text{overlap}$$

$$\text{and} \quad j \in \left\{ 0, 1, 2, \dots, \frac{T - T_g}{T_g \cdot \text{step}} \right\} \quad (3)$$

The  $j$ th gating block loudness is defined as:

$$l_j = -0.691 + 10 \log_{10} \sum_i G_i \cdot z_{ij} \quad (4)$$

For a gating threshold  $\Gamma$  there is a set of gating block indices  $J_g = \{j : l_j > \Gamma\}$  where the gating block loudness is above the gating threshold. The number of elements in  $J_g$  is  $|J_g|$ .

The gated loudness of the measurement interval  $T$  is then defined as:

$$\text{Gated loudness, } L_{KG} = -0.691 + 10 \log_{10} \sum_i G_i \cdot \left( \frac{1}{|J_g|} \cdot \sum_{J_g} z_{ij} \right) \text{LKFS} \quad (5)$$

A two-stage process is used to make a gated measurement, first with an absolute threshold, then with a relative threshold. The relative threshold  $\Gamma_r$  is calculated by measuring the loudness using the absolute threshold,  $\Gamma_a = -70$  LKFS, and subtracting 10 from the result, thus:

$$\Gamma_r = -0.691 + 10 \log_{10} \sum_i G_i \cdot \left( \frac{1}{|J_g|} \cdot \sum_{J_g} z_{ij} \right) - 10 \text{ LKFS}$$

where:

$$J_g = \{j : l_j > \Gamma_a\}$$

$$\Gamma_a = -70 \text{ LKFS} \quad (6)$$

The gated loudness can then be calculated using  $\Gamma_r$ :

$$\text{Gated loudness, } L_{KG} = -0.691 + 10 \log_{10} \sum_i G_i \cdot \left( \frac{1}{|J_g|} \cdot \sum_{J_g} z_{ij} \right) \text{LKFS}$$

where:

$$J_g = \{j : l_j > \Gamma_r\} \quad (7)$$