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Acoustics — Methods for calculating loudness —

Part 3: Moore-Glasberg-Schlittenlacher method

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Foreword

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This document was prepared by Technical Committee ISO/TC 43, *Acoustics*.

A list of all parts in the ISO 532 series can be found on the ISO website.

Any feedback or questions on this document should be directed to the user's national standards body. A complete listing of these bodies can be found at www.iso.org/members.html.

Introduction

Loudness and loudness level are two perceptual attributes of sound describing absolute and relative sensations of sound strength perceived by a listener under specific listening conditions. Due to inherent individual differences among people, both loudness and loudness level have the nature of statistical estimators characterized by their respective measures of central tendency and dispersion determined for a specific sample of the general population.

The object of this document is to specify a calculation procedure based on the physical properties of sound for estimating loudness and loudness level of sound as perceived by listeners with otologically normal hearing under specific listening conditions. This procedure seeks numbers that can be used in many scientific and technical applications to estimate the perceived loudness and loudness level of sound without conducting separate human observer studies for each application. Because loudness is a perceived quantity, the perception of which may vary among people, any calculated loudness value represents only an estimate of the average loudness as perceived by a group of individuals with otologically normal hearing.

This document describes a method for calculating the loudness of time-varying sounds from the input signal, which may differ for the two ears. This calculation method is based on Moore-Glasberg-Schlittenlacher loudness calculation algorithms[1] to [5]. The method allows calculation of two quantities:

a) The short-term loudness, which is the momentary loudness of a short segment of a sound, such as a word in a speech sound or a single note in a piece of music.

b) The long-term loudness, which is the loudness of a longer segment of sound, such as a whole sentence or a musical phrase.

For most everyday sounds, both the short-term loudness and the long-term loudness vary over time. The loudness of sounds with durations up to 2 s or 3 s is well predicted from the maximum value of the long-term loudness reached during presentation of the sound[4][6] to [8]. For long-duration stationary sounds, the long-term loudness based on the method described in this document is very close to the loudness determined using the method described in ISO 532-2[9]. Deviations can occur for sounds with strong amplitude fluctuations, such as noises with narrow bandwidth; for such sounds the calculated loudness is more accurate for this document than for ISO 532-2.

The method of loudness calculation described in this standard can be applied to signals of any duration. However, it does not directly give an output corresponding to the overall loudness impression of a sound scene or soundscape over a period of minutes, hours, or days, which is called the “overall loudness” in this standard. The output of the method of loudness calculation described in this standard can be post-processed to estimate the overall loudness of a sound scene.

NOTE Post-processing is outside the scope of this document, but some possible methods have been described[10] to [13].

This document describes the calculation procedure leading to estimation of the loudness and loudness level of time-varying sounds and provides executable computer programs. The software provided with this document is entirely informative and provided for the convenience of the user. Use of the provided software is not required for conformity with the document.

NOTE Equipment or machinery noise emissions/immissions can also be judged by other quantities defined in various International Standards (see e.g. ISO 1996-1[14], ISO 3740[15], ISO 9612[16], and ISO 11200[17]).

Acoustics — Methods for calculating loudness —

Part 3:
Moore-Glasberg-Schlittenlacher method

1 Scope

This document specifies a method for estimating the loudness and loudness level of both stationary and time-varying sounds as perceived by otologically normal adult listeners under specific listening conditions. The sounds may be recorded using a single microphone, using a head and torso simulator, or, for sounds presented via earphones, the electrical signal delivered to the earphones may be used.

The method is based on the Moore-Glasberg-Schlittenlacher algorithm.

NOTE 1 Users who wish to study the details of the calculation method can review or implement the source code which is entirely informative and provided with the standard for the convenience of the user.

This method can be applied to any sounds, including tones, broadband noises, complex sounds with sharp line spectral components, musical sounds, speech, and impact sounds such as gunshots and sonic booms.

Calculation of a single value for the overall loudness over the entire period of a time-varying signal lasting more than 5 s is outside the scope of this document.

NOTE 2 It has been shown that, for steady tones, this method provides a good match to the contours of equal loudness level as defined in ISO 226:2003[18] and the reference threshold of hearing as defined in ISO 389-7:2019[19].

2 Normative references

The following documents are referred to in the text in such a way that some or all of their content constitutes requirements of this document. For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments) applies.

IEC 60318-7, *Electroacoustics – Simulators of human head and ear – Part 7: Head and torso simulator for the measurement of sound sources close to the ear*

IEC 61672-1, *Electroacoustics - Sound level meters - Part 1: Specifications*

3 Terms and definitions

For the purposes of this document, the following terms and definitions apply.

ISO and IEC maintain terminology databases for use in standardization at the following addresses:

— ISO Online browsing platform: available at https://www.iso.org/obp

— IEC Electropedia: available at https://www.electropedia.org/

3.1
sound pressure level
L_p
ten times the logarithm to the base 10 of the ratio of the square of the sound pressure, p, to the square of a reference value, p_0, expressed in decibels

$$L_p = 10 \lg \frac{p^2}{p_0^2} \text{dB}$$

where the reference value, p_0, in air is 20 µPa

Note 1 to entry: Because of practical limitations of the measuring instruments, p^2 is always understood to denote the square of a frequency-weighted, frequency-band-limited or time-weighted sound pressure. If specific frequency and time weightings as specified in IEC 61672-1 and/or specific frequency bands are applied, this should be indicated by appropriate subscripts; e.g. $L_{p,\mathrm{AS}}$ denotes the A-weighted sound pressure level with time weighting S (slow). Frequency weightings such as A-weighting should not be used when specifying sound pressure levels for the purpose of loudness calculation using the current procedure.

Note 2 to entry: This definition is technically in accordance with ISO 80000-8:2020, 8-22[20].

3.2
filter
any device or mathematical operation which, when applied to a complex signal, passes energy of signal components of certain frequencies while substantially attenuating energy of signal components of all other frequencies

3.3
band-pass filter
filter (3.2) that passes signal energy within a certain frequency band and rejects most of the signal energy outside of this frequency band

3.4
sound spectrum
representation of the magnitudes (and sometimes of the phases) of the components of a complex sound as a function of frequency

3.5
auditory filter
filter (3.2) within the human cochlea describing the frequency resolution of the auditory system, whose characteristics are usually estimated from the results of masking experiments

3.6
***ERB*$_\mathbf{n}$**
equivalent rectangular bandwidth of the auditory filter for otologically normal persons
width of an idealised rectangular *band-pass filter* (3.3) that has the same peak transmission as the *auditory filter* (3.5) at the same centre frequency and that passes the same power for a white noise input (in Hz)

Note 1 to entry: The subscript n indicates that the value applies for listeners with otologically normal hearing.

Note 2 to entry: The unconventional use of a multiletter abbreviated term presented in italics and with a subscript is used here in the place of a symbol to maintain the use of an established notation and to avoid confusion.

3.7
***ERB*$_n$-number scale**
equivalent rectangular bandwidth number scale
transformation of the frequency scale constructed such that an increase in frequency equal to one ERB_n (Hz) (3.6) leads to an increase of one unit on the ERB_n-number scale

Note 1 to entry: The unit of the ERB_n-number scale is the Cam. For example, the value of ERB_n for a centre frequency of 1 000 Hz is approximately 132 Hz, so an increase in frequency from 934 Hz to 1 066 Hz corresponds to a step of one Cam. The equation relating ERB_n-number to frequency is given in 7.4.

3.8
loudness level
sound pressure level of a frontally incident, sinusoidal plane progressive wave, presented binaurally at a frequency of 1 000 Hz that is judged by otologically normal persons as being as loud as the given sound

Note 1 to entry: Loudness level is expressed in phons.

3.9
loudness
perceived magnitude of a sound, which depends on the acoustic properties of the sound and the specific listening conditions, as estimated by otologically normal listeners

Note 1 to entry: Loudness is expressed in sones.

Note 2 to entry: Loudness depends primarily upon the sound pressure although it also depends upon the frequency, waveform, bandwidth, and duration of the sound.

Note 3 to entry: One sone is the loudness of a sound whose loudness level is 40 phon.

Note 4 to entry: A sound that is twice as loud as another sound is characterized by doubling the number of sones.

3.10
short-term loudness
loudness of an individual brief segment of sound, such as a syllable in speech, a single musical note, or a short burst of a sound, typically lasting up to 500 ms

3.11
long-term loudness
loudness of a long sound, such as a whole sentence, a musical phrase, or a continuous noise, typically lasting up to 5 s

Note 1 to entry: The overall loudness of a sound or soundscape lasting longer than 5 s can be estimated by post-processing of the long-term loudness as a function of time. Such post-processing is outside the scope of this standard, but some possible methods are described in References [10] to [13].

3.12
excitation
E
output of an *auditory filter* (3.5) centred at a given frequency, specified in units that are linearly related to power

Note 1 to entry: An excitation of 1 unit is produced at the output of an auditory filter centred at 1 000 Hz by a tone with a frequency of 1 000 Hz with a sound pressure level of 0 dB presented in a free field with frontal incidence.

3.13
excitation level
L_E
ten times the logarithm to the base 10 of the ratio of the *excitation* (3.12) at the output of an *auditory filter* (3.5) centred at the frequency of interest to the reference *excitation* (3.12), E_0

$$L_E = 10 \lg \frac{E}{E_0} \text{ dB}$$

where the reference excitation E_0 is the excitation produced by a 1 000 Hz tone with a sound pressure level of 0 dB presented in a free field with frontal incidence

3.14
specific loudness
N'
calculated loudness evoked over a frequency band with a bandwidth of 1 ERB_n centred on the frequency of interest

4 General

The method described in this document specifies a method for calculating loudness and loudness level of any sound based on the Moore-Glasberg-Schlittenlacher procedure.

The method involves a sequence of stages. Each stage is described below. However, it is envisaged that those wishing to calculate loudness using this procedure will use one of the computer programs (see Annex A) provided with this document that implements the described procedure. It is not expected that the procedure will be implemented "by hand". Such computations would be very time consuming. The source code provided in Annex A gives an example of the implementation of the method. Other implementations using different software are possible.

NOTE 1 The computational procedure described in this document is an updated version of procedures published earlier elsewhere in References [1] to [5].

NOTE 2 Uncertainties are addressed in Clause 8.

5 Input signal

The signal that is used as input to the algorithm is the waveform for each ear (left and right), sampled using a 32 kHz sampling rate. If the Matlab®[1)] code described in Annex C is used, higher sampling rates for the signal are allowed. These are automatically converted by the Matlab® software to a 32 kHz sampling rate. The signal can be obtained in three ways.

5.1 Single microphone

The sound can be recorded using a single microphone placed at the centre of the position of the listener's head, after the listener has been removed from the sound field. In this case, the sound would be diotic (the same at the two ears) and the single recorded signal would be presented to both input channels of the algorithm.

5.2 Two microphones in the ear canals or microphones in a head and torso simulator

The sound can be recorded using two small probe microphones with the tips placed close to each ear drum (left and right) or using the two ear simulators (left and right) in a head and torso simulator.

5.3 Earphone presentation

If the sound is delivered via earphones, the input signals for the algorithm correspond to the electrical signals delivered to the earphones, but with allowance for the transfer function from each earphone to the eardrum; see 7.2.4.

1) Matlab® is a trademark of MathWorks. This information is given for the convenience of users of this document and does not constitute an endorsement by ISO of the product named. Equivalent products may be used if they can be shown to lead to the same results

6 Instrumentation

Measuring instrumentation used to acquire a signal to be used as an input for method 5.1 and 5.2 shall conform to IEC 61672-1. The microphone(s) used for method 5.1 shall have an omnidirectional characteristic or a free-field characteristic. If a head and torso simulator is used it shall conform to IEC 60318-7. For signals acquired using a head and torso simulator, the transfer function of the simulator as supplied by the equipment manufacturer or acquisition software shall be allowed for.

7 Description of the method

7.1 General

The procedure involves a sequence of processing operations, as illustrated in Figure 1.

For each ear, the processing operations are:

a) a filter to allow for the effects of transfer of sound through the outer and middle ear;

b) a short-term spectral analysis of the sound spectrum with greater frequency resolution at low than at high frequencies;

c) calculation of an excitation pattern, representing the magnitudes of the outputs of the auditory filters as a function of centre frequency;

d) application of a compressive nonlinearity to the output of each auditory filter to transform excitation to specific loudness;

e) smoothing over time of the resulting instantaneous specific loudness pattern using an averaging process resembling an automatic gain control (AGC) to give short-term specific loudness.

Subsequent stages are:

f) the short-term specific loudness patterns for each ear are used to calculate broadly-tuned binaural inhibition functions, the amount of inhibition depending on the relative short-term specific loudness at the two ears;

g) the inhibited specific loudness patterns are summed across frequency to give an estimate of the short-term loudness for each ear;

h) the binaural short-term loudness is calculated as the sum of the short-term loudness values for the two ears;

i) the long-term loudness for each ear is calculated by smoothing the short-term loudness for that ear, again by a process resembling AGC;

j) the binaural long-term loudness is obtained by summing the long-term loudness across ears.

These steps are described sequentially in 7.2 to 7.9.

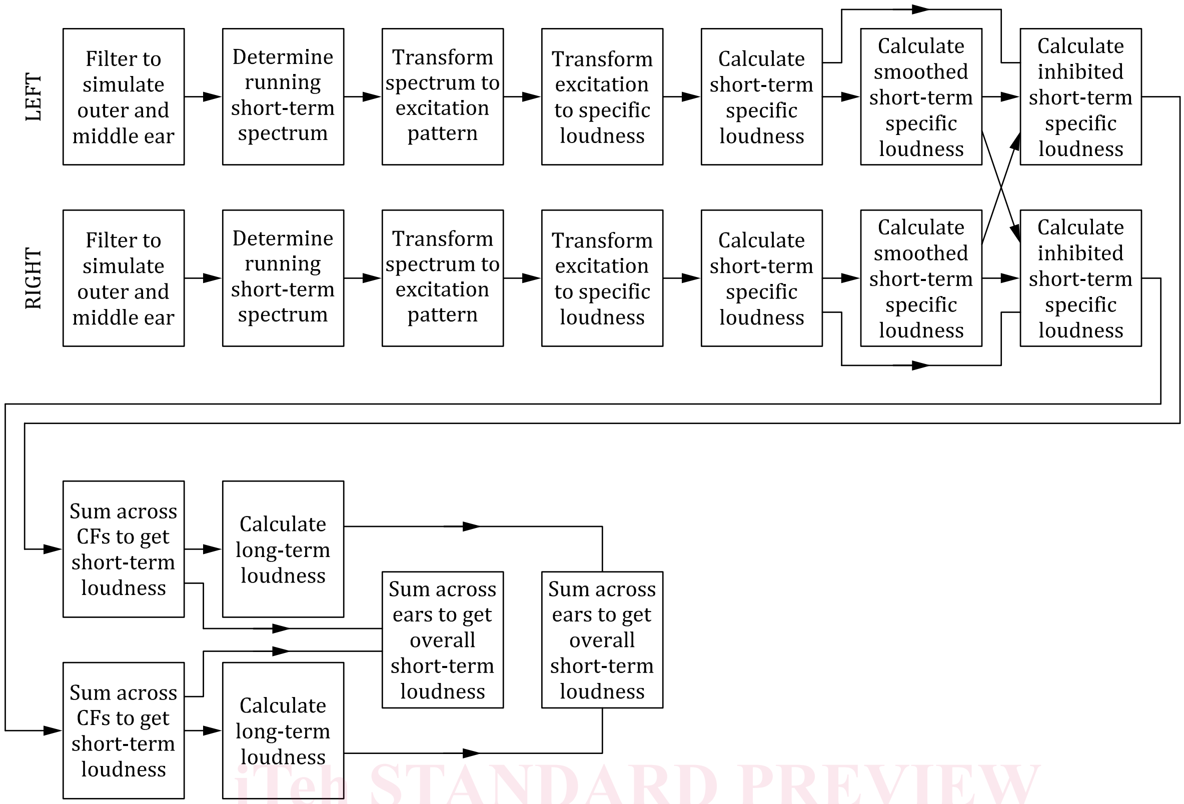

Figure 1 — Flow chart illustrating the sequence of processing operations in the method

7.2 Transfer of sound through the outer and middle ear

7.2.1 General

The transfer of sound through the outer and middle ear is modelled using one of three finite impulse response (FIR) filters with 4 097 coefficients. Different filters are used depending on the method by which the sound was picked up and the method by which the sound was delivered to the listeners. Each filter represents the combined effect of the outer ear and the middle ear. The transfer function for the middle ear is the same as for ISO 532-2:2017[9], 7.3 and is specified in column 4 of Table 1. This transfer function is referred to as “middle ear only”.

Table 1 — Transfer functions

Frequency Hz	Difference between the sound pressure level at the tympanic membrane and the sound pressure level measured in the free field (in the absence of a listener) dB	Difference between the sound pressure level at the tympanic membrane and the sound pressure level measured in the diffuse field (in the absence of a listener) dB	Scaled transfer function value for the middle ear dB
20	0,0	0,0	−39,6
25	0,0	0,0	−32,0
31,5	0,0	0,0	−25,85
40	0,0	0,0	−21,4
50	0,0	0,0	−18,5
63	0,0	0,0	−15,9

[a] Values are in a range that has not been validated.