



Predicting the loudness of non-stationary sounds: Zwicker's original envelope extraction vs. DIN 45 631/A1:2010

Florian Völk¹

Bio-Inspired Information Processing, Institute of Medical Engineering,
Technische Universität München, Boltzmannstraße 11, 85748 Garching, Germany

WindAcoustics UG (haftungsbeschränkt)
Mühlbachstraße 1, 86949 Windach, Germany

ABSTRACT

All currently standardized procedures for computationally predicting the loudness elicited by non-stationary sounds (under normal conditions in a typical subject) are based on the work of Zwicker and co-workers. The corresponding algorithms consequently share a spectral channel-based processing scheme, roughly mimicking peripheral auditory processing. With the same motivation, only the temporal envelopes in the channels are analyzed further. This contribution compares the envelope-extraction procedure proposed by Zwicker and co-workers to that of DIN 45631/A1:2010 regarding its intended purpose, implementation details, and effects on the predicted loudness. Based on the results, implications for future loudness-prediction standards are discussed.

Keywords: Loudness Models, Envelope Extraction I-INCE Classification of Subjects Numbers: 79.4, 63.1

1. INTRODUCTION

An algorithm for predicting the loudness of steady sounds perceived by a typical listener under normal conditions, derived from the loudness model of Zwicker and co-workers [1, 2, 3, 4], was standardized in DIN 45631 [5] and ISO 532-B [6]. More recently, a proposal from the same group applicable also for non-steady sounds [4, 7] was adapted by DIN 45631/A1 [8]. A similar algorithm for steady and non-steady sounds is designated to be included in the future ISO/DIS 532-1, the first part of the current revision of ISO 532 [9]. Consequently, Zwicker's procedure is gaining even more relevance and is considered an appropriate basis for predicting loudness and hearing sensations in general, in future standards and applications [e. g., 10, 11, 12].

Having said that, it appears reasonable to have a closer look at and discuss consistencies and differences of the original model and the standard. This is especially relevant with regard to extending the basic procedure beyond its current use for loudness and sharpness prediction [8, 13] towards a unified algorithm for predicting hearing sensations, as suggested for example by [12]. In this contribution, an early processing stage of the model, the envelope extraction in the spectral channels, is compared between the original proposals and DIN 45631/A1 [8]. Initially, the relevant physiological peripheral auditory processes are discussed briefly. Then, the basic structure of the original model is reviewed with a focus on the motivation and implementation of the envelope extraction, based on selected publications. Subsequently, the algorithm of the standard and typical implementations are reviewed. On that course, a comparison of the procedures and their implications is given. Eventually, the paper concludes with recommendations for future loudness prediction algorithms, which are relevant also for phenomenological auditory models in general.

2. RELEVANT ASPECTS OF PERIPHERAL AUDITORY PROCESSING

Zwicker-type loudness models were developed based on a mixture of psychoacoustical, physical, and physiological results [1, 7]. Especially regarding their initial stages, the resulting procedures closely follow the actual peripheral auditory processing [e. g., 4]: the sound field is physically altered frequency dependently in a sound-source location-specific manner by the body (mainly head, upper body, and outer ears), before being sampled by means of the sound-pressure time signals picked up by the eardrums.

¹email: voelk@tum.de, voelk@windacoustics.com

The eardrum vibrations are conducted mechanically by the middle-ear bones to the inner ear, entering the liquid-filled cochleae through the oval windows. Each cochlea contains, spanning its whole length and dividing its inside, a structure combined of basilar membrane, organ of Corti, and tectorial membrane. On the upper side of the organ of Corti sit, amongst others, the inner hair cells with stereociliae, bundles of little “hair”, at their upper ends. Oval-window vibrations propagate through the liquid inside the cochlea and cause the above-mentioned structure to vibrate (for a recent overview cf. e. g., [14]).

The mechanical and hydrodynamical properties of the assembly, especially of the basilar membrane, cause the vibrations to propagate as traveling waves along the basilar membrane. These traveling waves induce frequency-dependent locations of maximum vibration amplitudes of the structure, which are the reason of the often-quoted frequency-position transformation of the inner ear [4]. The system is further affected by efferent feedback (from higher processing stages on the path up towards the brain) and must therefore be considered an adaptive system [4].

The vibrations of the above-mentioned structure cause, in combination with the surrounding fluids, a relative motion of tectorial membrane and stereociliae. The stereocilia motion initiates processes which eventually transform the local mechanical vibrations to electrical signals on the locally attached nerve terminals, the nerve action potentials (spikes) conveyed further towards the brain [14]. The spike generation is not equally effective for both directions of stereocilia deflection: spikes are primarily generated on deflection towards the cochlear axis [15]. Furthermore, the speed of the spike generation process is limited per auditory nerve fiber to the order of hundreds per seconds, as an hair cell, after firing, needs a certain amount of time to recuperate before being able to fire again. Each inner hair cell is connected to some ten afferent nerve fibers transporting information towards the brain, providing redundancy and probably also increasing temporal accuracy somewhat. At least at frequencies below about 1 kHz, the firing occurs in phase with the stereocilia deflection [16].

A simple engineering model of the above described system with frequency-selective behavior (frequency-position transformation; ignoring time variances introduced by the adaptive system) is an adequately parametrized bank of K bandpass filters (FB), operating on the frequency-dependently attenuated (index a) sound pressure $p_a(t)$. The peripheral (PE) attenuation $a_{PE}(f)$ accounts for sound-field, body, and middle-ear influences on the sound-pressure signal $p(t)$, as schematically shown in the left part of figure 1.

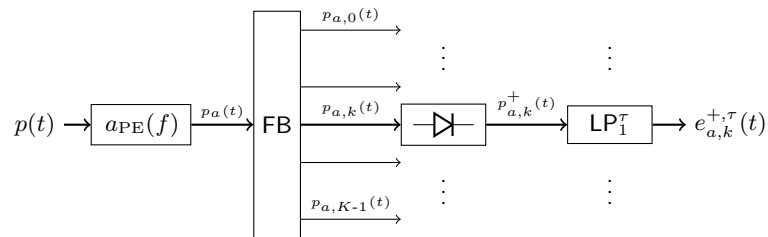


Figure 1 – Simplified engineering model of peripheral auditory processing: the sound-pressure-time signal $p(t)$ is frequency dependently attenuated ($a_{PE}(f)$ reflects influences of sound field, body, and middle ear) and filtered by a bank of K bandpass filters (FB). The filter-output signals $p_{a,k}(t)$ are rectified (diode) and lowpass filtered (LP_1^τ : first-order lowpass with time constant τ), resulting in the envelopes $e_{a,k}^{+, \tau}(t)$.

From an engineering point of view, a diode (one-way rectifier) followed by a first-order lowpass filter (LP_1^τ) can be considered a rough model of the information-reduction processes encoding the nerve signals as described above (right part of figure 1). In terms of the effect on the bandpass-filter-output signals $p_{a,k}(t)$, the diode-lowpass combination is an amplitude-modulation demodulator, extracting the signal envelopes $e_{a,k}^{+, \tau}(t)$, while somewhat suppressing higher-frequency components.

3. RELEVANT ASPECTS OF ZWICKER-TYPE LOUDNESS MODELS

The most recent outline of Zwicker-type loudness models in general can be found in Fastl and Zwicker (2007) [4, sec. 8.7]. On page 237, the authors describe the model structure shown by the block diagram in their figure 8.26. In the corresponding text, relevant aspects of the envelope extraction are given as follows: “The sound pressure time function $p(t)$ is picked up by the microphone, fed to an amplifier and to a filter appropriate for free versus diffuse sound field. Then follows a filterbank, a rectifier and a lowpass with 2 ms time constant producing the temporal envelope of the filter outputs.” A similar description was given by Zwicker 1977 [7]: “an electrical network simulating this behavior should produce the envelope of the sound pressures falling in the [...] bands by means of [...] a square-law rectifier, [and] a lowpass filter with a time constant of 2 ms.” Apart from the physiological motivation of the model structure discussed

in the preceding section, the parameterization is likely motivated by psychoacoustic results, suggesting that the “characteristic duration of approximately 2 ms indicates the limit up to which the hearing system is [...] capable of evaluating the temporal structure of a sound” (Zwicker 1974, [17]).

Figure 2 shows a flowchart of the procedure extracting the envelope $e_{a,k}^{+2,2\text{ms}}(t)$. The pre-processing accounting for the different soundfields as discussed above is indicated here by the attenuation $a_{\text{SF}}(f_k)$.

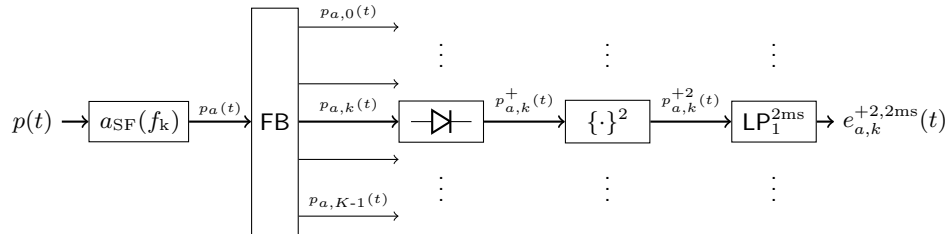


Figure 2 – Envelope extraction of Zwicker-type loudness models: the sound-pressure-time signal $p(t)$ is frequency dependently attenuated by $a_{\text{SF}}(f_k)$ and filtered by a bank of K bandpass filters (FB). The filter-output signals $p_{a,k}(t)$ are rectified (diode), squared ($\{\cdot\}^2$), and lowpass filtered ($\text{LP}_1^{2\text{ms}}$: first-order lowpass with time constant $\tau = 2$ ms), resulting in the envelopes $e_{a,k}^{+2,2\text{ms}}(t)$.

Summarizing, Zwicker-type loudness models closely follow the procedure outlined by figure 1, with the parameter $\tau = 2$ ms, resulting in the lowpass-filter bandwidth $\Delta f = 1/(2\pi\tau) \approx 80$ Hz. Additionally, the filterbank-output signals $p_{a,k}(t)$ are being squared after one-way rectification (“square-law rectifier”, [7]).

4. RELEVANT ASPECTS OF DIN 45631/A1:2010

In section B.1, DIN 45 631/A1:2010 [8], entitled “Calculation of loudness [...] – Zwicker method [...]”, states that “in the following, procedures are described, which simulate the temporal processing of sounds in the human hearing system during judging the loudness of non-stationary sounds” (translated from the German original). Section B.4.1 of the standard gives the “Exemplary structure: after amplification of the microphone signal [...], free or diffuse sound field is selected [...]. Then follows a third-octave filter bank according to DIN EN 61260, a rectification, and lowpass filtering. The time constant τ is selected frequency dependently”, as a function of the third-octave-center frequency f_k of the respective channel, according to

$$\tau_k = \begin{cases} 2/(3f_k) & \text{if } f_k \leq 1 \text{ kHz,} \\ 2/(3 \text{ kHz}) = (2/3) \text{ ms} & \text{otherwise.} \end{cases} \quad (1)$$

The structure is outlined in figure 3 and discussed in the following.

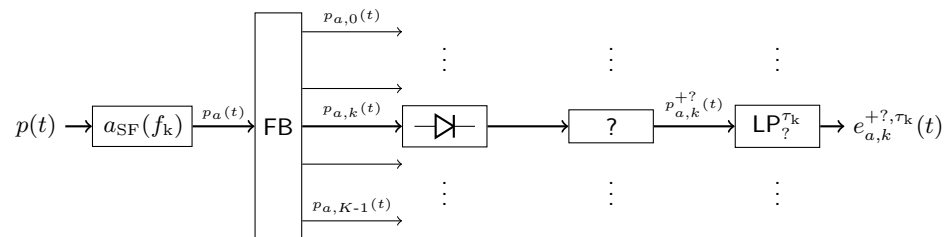


Figure 3 – Envelope extraction of DIN 45631/A1:2010 [8]: the sound-pressure-time signal $p(t)$ is frequency dependently attenuated by $a_{\text{SF}}(f_k)$ and filtered by a bank of K bandpass filters (FB). The filter output signals $p_{a,k}(t)$ are rectified, with unspecified compression/expansion, and lowpass filtered ($\text{LP}_?^{\tau_k}$: lowpass of unspecified order with time constant τ_k), resulting in the envelopes $e_{a,k}^{+?,\tau_k}(t)$.

4.1. Rectification

Unfortunately, DIN 45 631/A1:2010 [8] specifies only that the signal in each spectral channel must be processed by “rectification”. This leaves room for interpretation, not explicitly requiring one-way or two-way rectification and not explicitly excluding compression. However, the block diagram shown in the normative appendix B of the standard [8, figure B.1] is content-wise identical to Fig. 8.26 of Fastl and Zwicker (2007) [4], containing a diode-lowpass combination in each channel, which could be assumed to

indicate (in line with Fastl and Zwicker, [4]) a square-law one-way rectification. As discussed above, the physiological processes in the human auditory system suggest one-way rectification.

Two-way square-law rectification could for example be implemented by squaring the filter-output time signals $p_{a,k}(t)$, resulting in $p_{a,k}^2(t)$. To the author's best knowledge, there is no auditory incentive for two-way rectification. However, future standardization appears to be planned based on two-way rectification: "The output signals of the filter array are squared and smoothed by [...] low-pass filters" [18]. The latter proposal might probably have been evolved from older literature, as Zwicker and Feldtkeller (1967) described in their book "Das Ohr als Nachrichtenempfänger" [22, sec. 77] the envelope extraction of a loudness-calculation procedure implemented on an analog calculator (p. 200, translated from the German original): "The voltage in each channel is being squared (root-mean-square value) and routed to a RC network, which correctly weights the impulsiveness of the sounds." However, this proposal might have been motivated more by the limited technical capabilities at the time and/or the lack of the respective research results, and was later revised in the succeeding book "Psychoakustik" (Zwicker 1982) [23, sec. 15.4, p. 143]. Translated from the German original, the corresponding section reads: "Each bandpass [...] is followed by a rectifier, [and] a lowpass" [23, figure 15.15]. All later versions, including the most recent (Fastl and Zwicker 2007, [4]) specify "a filterbank, a rectifier and a lowpass with 2 ms time constant".

In order to illustrate the implications of the specific rectification method, figure 4 illustrates both discussed possibilities for an exemplary third-octave channel centered at $f_k = 125$ Hz, in response to a 100% amplitude-modulated pure tone with a frequency of 125 Hz (modulation frequency 4 Hz; all computations and simulations shown in this paper were carried out using [19]).

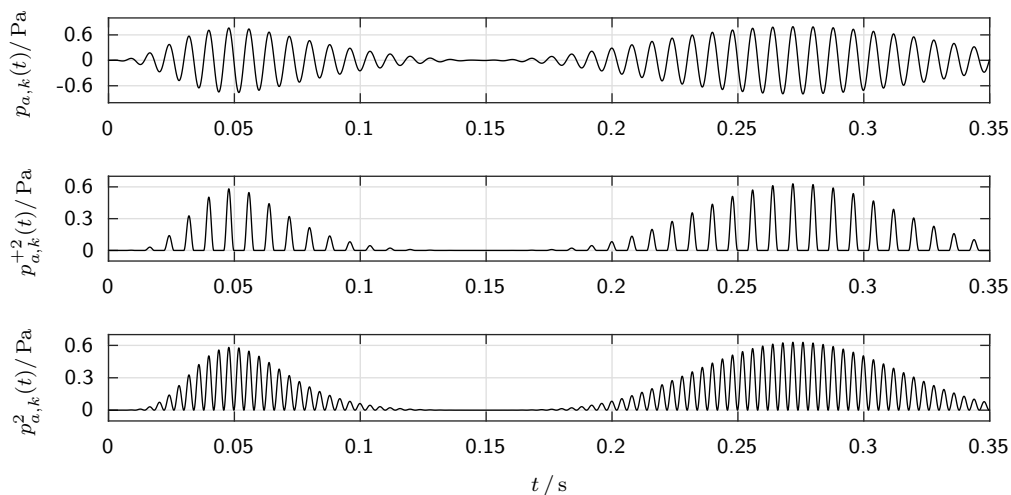


Figure 4 – Square-law rectification illustrated using a 100% amplitude-modulated pure 125 Hz tone (4 Hz modulation frequency). Output signal of a third-octave filter centered at 125 Hz (upper panel), one-way-rectified signal (middle panel), and two-way rectified signal (bottom panel).

The upper panel of figure 4 shows the filter-output signal $p_{a,k}(t)$, the middle panel its one-way square-law rectified version $p_{a,k}^{+2}(t)$, and the lower panel the two-way square-law rectified version $p_{a,k}^2(t)$. Comparing the middle and lower panels to the upper panel clearly indicates that two-way rectification (squaring, bottom panel) doubles the signal frequency in the channel, in contrast to one-way rectification (middle panel), which preserves the original frequency. Regarding the accuracy of loudness-prediction results, preserving the original frequency is especially important as the auditory system is able to identify the current position within the period of signals with low-frequency content [20]. For loudness prediction, however, the overall loudness after spectral summation [cf. 5] only contains the low-frequency effects correctly if the bandpass-filter delays sufficiently reflect the corresponding physiologic delays.

4.2. Lowpass Filtering

Regarding the lowpass filter following the rectification, DIN 45 631/A1:2010 [8] specifies only the frequency-dependent time constant τ_k given by equation 1 above, not an order (slope steepness) or a filter structure/implementation. As all the mentioned parameters influence the resulting signal envelope $e_{a,k}^{+?,\tau_k}(t)$, their influences will be discussed. Figure 5 shows, as a function of frequency, the lowpass-filter bandwidths $\Delta f = 1/(2\pi\tau)$ corresponding to the time constants τ_k according to DIN 45 631/A1:2010 ([8],

dash-dotted dark) and $\tau = 2$ ms as proposed by Zwicker ([4, 7], solid dark). For comparison purposes, the frequency-dependent third-octave bandpass-filter bandwidth is also depicted ([21], solid gray).

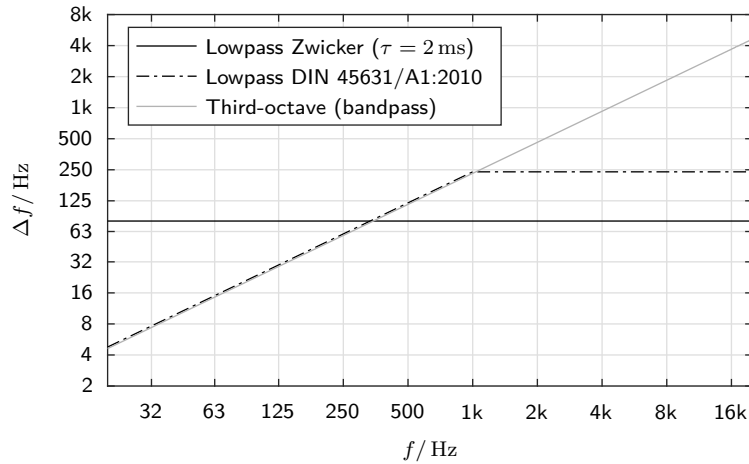


Figure 5 – Lowpass-filter bandwidths $\Delta f = 1/(2\pi\tau)$ corresponding to the time constants τ_k according to DIN 45 631/A1:2010 ([8], dash-dotted dark) and $\tau = 2$ ms as proposed by Zwicker ([4, 7], solid dark). The solid light contour indicates the third-octave bandpass-filter bandwidth.

Visual inspection reveals that τ_k is defined by DIN 45 631/A1:2010 [8] so that the corresponding lowpass-filter bandwidth equals the third-octave bandwidth at center frequencies $f_k < 1$ kHz (dash-dotted dark and solid gray contours in figure 5). While a motivation for setting $\tau_k = (2/3)$ ms at higher frequencies is not apparent to the author, the design choice at lower frequencies may go back to the fact that, as noted by Zwicker 1977 [7], “the rise and decay times [within the bandpass-filter channels] are limited only by the filter bandwidths at low frequencies and by the 2-ms time constant at high frequencies.” This notion may have resulted in the erroneous assumption that, at low frequencies, the (lowpass) time constants τ_k can be enlarged to correspond to the respective (bandpass-)filter bandwidths without consequences. It is especially important to recognize that between the bandpass and lowpass filters, a non-linear rectification takes place, which does not possess the commutative property of linear systems.

Figure 6 illustrates, by the example of a 100% amplitude-modulated pure 125 Hz tone (4 Hz modulation frequency), the effect of different time constants and filter orders on the time signal in a low-frequency channel (third-octave filter centered at 125 Hz).

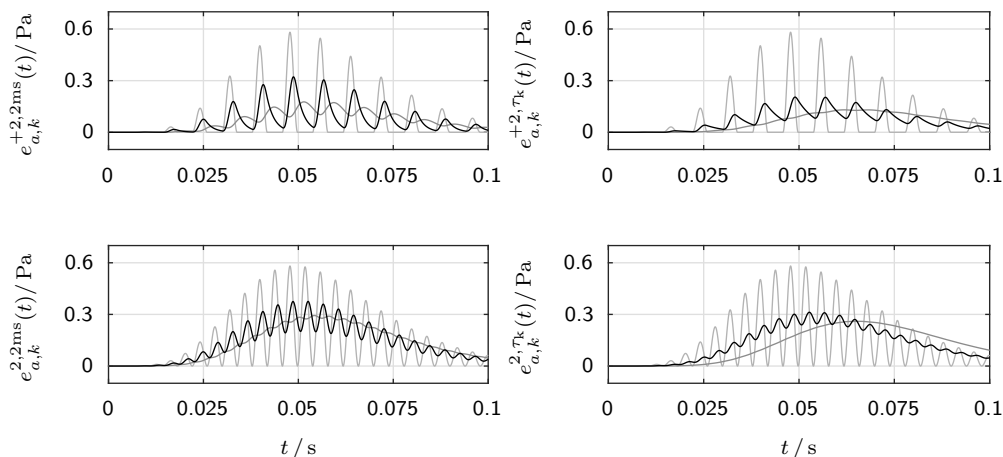


Figure 6 – Envelope extraction illustrated using a 100% amplitude-modulated pure 125 Hz tone (4 Hz modulation frequency) by the example of the output signal of a third-octave filter centered at 125 Hz. First-order (bold dark) and third-order (thin dark) lowpass filtering with $\tau = 2$ ms (left column) and τ_k (according to equation 1, right column) of the one-way-rectified signal (upper row) and the two-way rectified signal (bottom row). Envelopes (dark) and rectified signals (light, reprinted from figure 4).

The upper row of figure 6 shows the one-way square-law-rectified filter-output signals (light) and corresponding envelopes (dark), while the bottom row shows the same data for the two-way rectified signals. The light contours are reprinted from figure 4. In both rows, the lowpass filters for the envelope extraction were parameterized with $\tau = 2$ ms in the left panels and τ_k according to equation 1 on the right. As the aforementioned plans for future loudness standards include a combination of the frequency-dependent time constant τ_k with third-order lowpass filters (“The output signals of the filter array are squared and smoothed by third-order low-pass filters [... with ...] frequency-dependent time constant” [18]), first-order as well as third-order lowpass filtering is included (bold dark and thin dark, respectively).

Taking into account that the auditory system is (monaurally) able to resolve the temporal fine-structure of signals in the depicted low-frequency range [20], a loudness-prediction algorithm should resolve this as well. As discussed in the previous section, the two-way rectified (squared) signals (bottom row) alter the fine structure by doubling the signal frequency. If these two-way rectified signals are additionally temporally smoothed by applying a lowpass filter, the fine-structure information is falsified more. In the extreme case of third-order lowpass filtering the squared signal with the time constant τ_k (as proposed by [18]; thin dark contour in the lower right panel in figure 6), all fine-structure information is lost. Zwicker’s parametrization (bold dark contour in the upper left panel) on the contrary clearly preserves the temporal fine structure. As mentioned above, for being reflected correctly in the overall loudness, the bandpass-filterbank delays must correspond sufficiently well to the modeled physiological effects.

4.3. Sound-field attenuation

While not being within the primary focus of this study, the attenuation $a_{SF}(f_k)$ accounting for the difference between free and diffuse sound fields (cf. figure 2) is addressed in the following. DIN 45 631/A1:2010 [8] specifies in section B.4.1 that “after amplification of the microphone signal [...], free or diffuse sound field is selected (see appendix A, lines 1 660 to 1 710). Then follows a bank of third-octave filters”. However, appendix A (part of DIN 45631:1991 [5]) only specifies third-octave-band level differences (DDF[k]). If their effect is to be applied to the sound-pressure time signal $p(t)$ before the filterbank, as requested by [8, sec. B.4.1 and figure B.1], a filter must be constructed to account for its effect. This filter is not specified by [8]. Additionally, no changes were made to appendix A [5] regarding the level differences DDF[k]. However, if their effect is already applied to $p(t)$, the level differences must be removed from the subsequently applied procedure described in appendix A.

5. SUMMARY AND RECOMMENDATIONS FOR FUTURE STANDARDS

Zwicker-type loudness models recommend, for computationally predicting the loudness of non-stationary sounds, a spectral channel-based processing scheme, extracting the signal envelopes in the channels by one-way square-law rectification followed by first-order lowpass filtering with a time constant of 2 ms. This strategy roughly mimics peripheral auditory processing. While, according to its title, implementing Zwicker’s procedure, DIN 45631/A1:2010 [8] specifies a different, frequency-dependent time constant and leaves the rectification-law and filter-order undefined. Recent reports on activities towards future standardization of loudness-calculation procedures [18] indicate that the procedure of DIN 45631/A1:2010 is intended to be combined with two-way rectification and third-order lowpass filtering.

This contribution compares both procedures with regard to the underlying physiological processes and selected psychoacoustic data. The results suggest that Zwicker’s parameters are better suited to reflect both, physiological processes and perceptual results. It is therefore recommended to adopt Zwicker’s original procedure for future standards, and to describe the procedure meticulously and in sufficient detail. Proposed specifications and modifications of DIN 45631/A1:2010 are:

- Specification of one-way square-law rectification for the envelope extraction.
- Specification of first-order lowpass filtering for the envelope extraction.
- Switching to the frequency-independent time constant $\tau = 2$ ms for the envelope extraction.
- Specification of a filter accounting for the difference between free and diffuse sound fields, while removing the corresponding level difference from appendix A (for non-stationary sounds).
- Adaptation of figure B.1 to the actually described procedure.

The proposed modifications are especially relevant with regard to extending the basic procedure beyond its current use for loudness and sharpness prediction [8, 13] towards a unified algorithm for predicting hearing sensations, as suggested for example by [12].

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