Providence, RI
NOISE-CON 2016
2016 June 13-15

Computational Loudness Prediction: Ambiguities and Potential Improvements of ANSI S3.4-2007

Florian Völk [1,2]
[1] Bio-Inspired Information Processing
Technical University of Munich
Garching, Germany
[2] WindAcoustics UG, Windach, Germany
voelk@windacoustics.com

Jesko L. Verhey [3]
[3] Department of Experimental Audiology
Otto von Guericke University
Magdeburg, Germany

ABSTRACT
The current American national standard for computationally predicting the loudness of steady sounds, ANSI S3.4-2007, is accompanied by an informative software program implementing the procedure. This contribution suggests some mathematical formulations for parameters of the method for calculating loudness, with a special focus on clarifying potential ambiguities of the standard. The proposed changes may contribute to a more specific and thereby improved next-generation ANSI loudness standard.

1 INTRODUCTION
According to the foreword, the procedure for the computational prediction of loudness recommended by ANSI S3.4 [1] is based on a method developed from the loudness model of Zwicker and co-workers [2, 3, 4, 5] by Moore et al. [6, 7]. Section 3.1 of the standard [1] states that the "procedure is similar, but not identical, to that" of [6, 7]. In the following, an explicit distinction between the procedures is made only where necessary, otherwise both are referred to as the "ANSI model."

An algorithm directly derived from and more similar to the original procedure of Zwicker and co-workers was standardized for steady sounds in DIN 45 631 [8] and ISO 532 [9, B], and for non-steady sounds in DIN 45 631/A1 [10]. A comparable algorithm applicable to steady and non-steady sounds will be included in the future ISO/DIS 532-1, the first part of the current revision of ISO 532 [9]. The second part of this revised version of the international standard, ISO/DIS 532-2, will be applicable for steady sounds and will contain a revision of ANSI S3.4 [1].

While the question of the best choice for future loudness-prediction standards has been addressed elsewhere [e. g. 11, 12, 13], the aim of this contribution is to revisit the definitions of parameters and procedures of ANSI S3.4-2007 [1], to address possible inconsistencies, and to propose potential improvements. The paper is structured as follows: initially, the model structure is briefly reviewed. In a second step, the parameter definitions (graphical representations, standardized tables, and textual components) are evaluated regarding the best-suited definition of the loudness-prediction procedure. The new approach is initially validated by comparison to a custom implementation of the standard, to tabulated values of the standard document, and to predictions of the software provided with ANSI S3.4 [1].

2 MODEL STAGES AND OVERVIEW

The loudness prediction with the ANSI model is considered a two-stage procedure here, with the stages characterized by their specific and for most conditions different number of spectral channels. The channel distribution of the first stage, referred to as "pre-processing" here, is determined by the user-specified intensity-density spectra representing the model input (various options are permissible, as defined by [1, sec. 3.2]). The channel distribution of the second stage, referred to as the actual "loudness prediction" here, is defined by the procedure and is therefore independent of the way the user specifies the model input. Comparable requirements apply to the parameters: while the center frequency and therefore the parameters of each pre-processing channel must be adapted to the specific user-selected spectral configuration, the parameters of the loudness prediction are only required for a finite number of given center frequencies. As those are uniquely defined, the parameters can be specified accordingly, for example by formulae or tables.

The procedure of [6, 7] is described using, amongst others, the only graphically-specified parameters sound-field-to-eardrum attenuation, middle-ear attenuation, excitation level at threshold, α, and A. ANSI S3.4 [1] specifies these parameters, partly modified, by tabulating specific discrete sample values, which are also shown graphically. As the algorithm requires intermediate values and possibly values beyond the tabulated range, interpolation and extrapolation procedures are defined additionally [1, sec. 3.3.4]. The parameters of the pre-processing stage are only discussed briefly in the following section. The main focus of this contribution is on the parameters of the second stage, the actual loudness prediction, which are discussed in section 4.

3 PRE-PROCESSING

The ANSI model provides several ways of specifying the physical model input, all of which are essentially followed by a conversion of the respective input to intensity-density spectra, here denoted $l_{\mathrm{IN}}(f_k)$ at K discrete center frequencies f_k, with $k = 0, \ldots, K-1$, within the respective frequency ranges of interest [cf. 1, sec. 3.2]. Zero intensity density is assumed outside these frequency ranges.

The spectra $l_{\mathrm{IN}}(f_k)$ are modified by two attenuation functions, one approximating the sound-field-to-eardrum or headphone-to-eardrum attenuation, respectively [1, sec. 3.3], here referred to as $a_{\mathrm{SF}}(f_k)$. The second mimics (according to [1, sec. 3.4]) aspects of the peripheral auditory processing and is here referred to as $a_{\mathrm{ME}}(f_k)$. Both attenuations can be summarized per component, resulting in $a_{\mathrm{PP}}(f_k) = a_{\mathrm{SF}}(f_k) + a_{\mathrm{ME}}(f_k)$. The pre-processing result is then given as the weighted intensity-density spectrum

$$l_{\mathrm{PP}}(f_k) = l_{\mathrm{IN}}(f_k) + a_{\mathrm{PP}}(f_k) = l_{\mathrm{IN}}(f_k) + a_{\mathrm{SF}}(f_k) + a_{\mathrm{ME}}(f_k). \tag{1}$$

ANSI S3.4-2007 [1] lists values for $a_{\mathrm{ME}}(f_k)$ and $a_{\mathrm{SF}}(f_k)$ at specific f_k, the latter for the free and diffuse sound fields, in tables 1, 2, and 3, combined with an interpolation procedure [1, sec. 3.3.4]. Headphone presentation can be accounted for by adjusting $a_{\mathrm{SF}}(f_k)$ according to the specific headphone's transfer characteristics [1, sec. 3.3.3].

4 LOUDNESS PREDICTION

In line with Zwicker's model [2], the weighted intensity-density spectrum resulting from the pre-processing (equation 1) is transformed to an excitation pattern $E(f_m)$, $m = 0, \ldots, M-1$, assumed to roughly correspond to the average physical excitation in the cochlea. Therefore, the intensity-densities $l_{\mathrm{PP}}(f_k)$ are summed for all k within the frequency-dependent critical-band around each center frequency f_m (for the definitions and nomenclature used in this paper cf. [14]). Implementing this idea, the ANSI model defines a procedure based on the level-dependent equivalent-rectangular bandwidth [1, sec. 3.5].

The last paragraph in section 3.5 of ANSI S3.4 [1] defines that standard-conform results are calculated at critical-band rates (in the standard denoted "ERB$_{\mathrm{N}}$-numbers", also cf. [14]) in steps of 0.1 between 1.8 and 38.9, calculated according to equation 4 of [1]. Using the inverse of this equation 4, the critical-band rates ("ERB$_{\mathrm{N}}$-numbers") correspond to $M = 372$ center frequencies f_m between $f_0 \approx 49\,\mathrm{Hz}$ and $f_{M-1} \approx 14.9\,\mathrm{kHz}$, with $m = 0, \ldots, M-1$. Comparably, the model described by Moore et al. [6, 7] employs center frequencies in the range from 50 Hz to 15 kHz ([6, sec. 1.4] and [7, sec. C]).

Zwicker's proposal [2] uses the excitation pattern $E(f_m)$ for calculating the specific-loudness pattern $N'(f_m)$. In the ANSI model, this is realized by different equations for different excitation ranges [1, eqs. 6, 7, 8]. While these formulae are not within the focus of this paper, they depend on parameters [1, sec. 3.6], which are discussed in the following.

4.1 Parameter "Excitation at absolute threshold"

A fundamental parameter of the ANSI model, which indirectly defines other parameters (cf. the following sections), is the excitation level at absolute threshold for monaural listening

$$L_{\mathrm{E_{THQ}}}(f_m) = 10\log_{10}\Big(E_{\mathrm{THQ}}(f_m)/E_0\Big)\,\mathrm{dB} = 10\log_{10}\Big(E'_{\mathrm{THQ}}(f_m)\Big)\,\mathrm{dB}, \quad (2)$$

with $E'_{\mathrm{THQ}}(f_m) = E_{\mathrm{THQ}}(f_m)/E_0$. E_0 is described as the excitation elicited by a 1 kHz pure tone presented at 0 dB SPL [1, sec. 3.5]. In general, normalized excitations $E' = E/E_0$ are indicated here by an apostrophe.

The ANSI model assumes the constant high-frequency excitation level at absolute threshold for monaural listening

$$L_{\mathrm{E_{THQ}}}(f_m \geq 500\,\mathrm{Hz}) = L_{\mathrm{E_{THQ,500\,Hz}}} = 3.73\,\mathrm{dB}, \quad (3)$$

corresponding, according to the standard [1, sec. 3.6.2], to the normalized excitation

$$E'_{\mathrm{THQ}}(f_m \geq 500\,\mathrm{Hz}) = E'_{\mathrm{THQ,500\,Hz}} = 2.36. \quad (4)$$

Excitation levels $L_{\mathrm{E_{THQ}}}(f_m)$ for $f_m < 500\,\mathrm{Hz}$ are defined by tabulated sample values in combination with an interpolation/extrapolation procedure ([1, tab. 4] with [1, sec. 3.6.2] and [1, sec. 3.3.4]). However, as the frequencies f_m for $L_{\mathrm{E_{THQ}}}(f_m)$ are exactly defined (see above), a direct definition, for example by tabulating the values at exactly those frequencies or by a formula, is possible and would be less susceptible to implementation influences. As an unambiguous and most exact definition is desirable for standardization purposes, this study proposes the formula

$$L_{\mathrm{E_{THQ}}}(f_m) = \begin{cases} 965\,\mathrm{dB}\ (f_m/\,\mathrm{Hz})^{-0.898} & \text{if } f_m \leq 486\,\mathrm{Hz}, \\ 3.73\,\mathrm{dB} & \text{otherwise.} \end{cases} \quad (5)$$

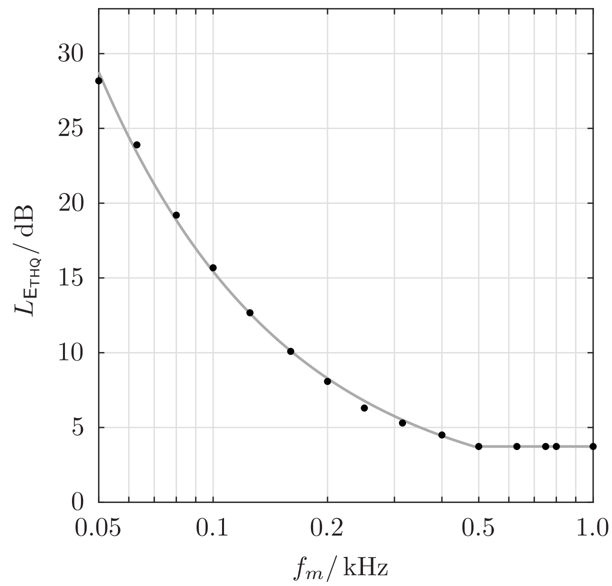

Figure 1: *Excitation level at absolute threshold for monaural listening $L_{E_{THQ}}(f_m)$, as used by ANSI S3.4-2007 [1], depicted as a function of analysis-center frequency f_m. The black dots represent the standardized data [1, table 4], the gray contour indicates equation 5 proposed here.*

As shown in figure 1, the results of equation 5 deviate slightly (on average by ± 0.18 dB and in every case less than 0.59 dB) from the tabulated data ([1, tab. 4], cf. also [1, fig. 6]). Taking the limited accuracy (inter-individual spread) and precision (intra-individual repeatability) of average pure-tone absolute-threshold levels [e. g. 15, 16] as an indicator for accuracy and precision of $L_{\mathrm{E_{THQ}}}(f_m)$, the above deviations appear tolerable; especially as they come with the advantage of a well-defined, continuous, and exponentially-decaying function of frequency. Furthermore, the modifications hardly affect the predicted loudness (less than 10% near threshold at the lowest audible frequencies, also cf. section 5). Consequently, equation 5 is recommended for a less ambiguously defined loudness-prediction standard.

4.2 Parameter G

The normalized excitation at absolute threshold $E'_{\mathrm{THQ}}(f_m)$ discussed in the previous section is further used in ANSI S3.4 [1] to define the parameter $G(f_m)$, referred to as the "low-level gain of the cochlear amplifier", and in consequence also the parameters $A(f_m)$ and $\alpha(f_m)$ [1, sec. 3.6.3]. While $A(f_m)$ and $\alpha(f_m)$ are tabulated as discrete samples [1, tabs. 5 and 6], the "low-level gain of the cochlear amplifier" is defined somewhat ambiguously in the running text [1, sec. 3.6.3]. The parameter $G(f_m)$ described in section 3.6.3 of ANSI S3.4-2007 [1] can be formulated mathematically as

$$G(f_m) = \frac{E'_{\mathrm{THQ}}(f_m \geq 500\,\mathrm{Hz})}{E'_{\mathrm{THQ}}(f_m)} = \frac{E'_{\mathrm{THQ,500\,Hz}}}{E'_{\mathrm{THQ}}(f_m)}, \tag{6}$$

with $E'_{\mathrm{THQ,500\,Hz}} = 2.36$, as given by equation 4 above. Replacing the textual definition of $G(f_m)$ in the current ANSI S3.4 [1] by equation 6 proposed here would provide an unambiguous definition without unnecessary conditions and would thereby improve the clarity of the description of the procedure.

Unfortunately, $G(f_m)$ is not only referred to as a weighting function in the standard [1, sec. 3.6.3], but also as an attenuation function (in dB, [1, tabs. 5 and 6]). This ambiguity can be overcome by restricting the definition of $G(f_m)$ to the linear weighting function and use a new term, e. g., $L_\mathrm{G}(f_m) = 10\log_{10} G(f_m)\,\mathrm{dB}$ as in this study, for the attenuation.

4.3 Parameter α

Table 5 of ANSI S3.4 [1] lists the sample values that, combined with the interpolation given by [1, sec. 3.3.4] and the textual description of [1, sec. 3.6.4], define the parameter $\alpha(f_m)$. A graphical representation of $\alpha(f_m)$ as a function of $L_\mathrm{G}(f_m)$ is given by figure 7 of [1]. To simplify the description of the calculation procedure, the formula

$$\alpha(f_m) = 0.113 + 0.087\,G(f_m)^{-0.099} = 0.113 + 0.087\left(\frac{E'_{\mathrm{THQ,500\,Hz}}}{E'_{\mathrm{THQ}}(f_m)}\right)^{-0.099} \tag{7}$$

is proposed here. Equation 7 represents the tabulated $\alpha(f_m)$ as a function of $L_\mathrm{G}(f_m)$ [1, tab. 5] on average within ± 0.0003, and with a maximum deviation of 0.0013. The relative deviations are less than 0.2% on average and below a maximum of approximately 0.7%.

Deviations of the above magnitude appear tolerable for the purpose of loudness prediction (cf. section 5), especially for being accompanied by the advantage of a well-defined, continuous, and exponentially-decaying function, without outliers. For implementations, this study recommends the second term of equation 7, since it directly relates $\alpha(f_m)$ to $E'_{\mathrm{THQ}}(f_m)$, without relying on the separate variable $G(f_m)$.

Compared to the current standard [1], equation 7 removes unnecessary conditions and an interpolation with implementation margins, and thereby again improves the unambiguity of the procedure. Because $\alpha(f_m)$ as well as $A(f_m)$ discussed in the following section appear to be derived empirically [6, 7], slight modifications that should not violate the relation to experimental data are considered tolerable.

4.4 Parameter A

The last parameter required for the loudness prediction with the ANSI model [1], $A(f_m)$, is defined comparably to $\alpha(f_m)$ by a combination of sample values at specific $L_\mathrm{G}(f_m)$ in table 6, the interpolation given by section 3.3.4, and the description in section 3.6.4 of the standard. Figure 8 of ANSI S3.4-2007 [1] is intended to visualize $A(f_m)$ as used in the standard. Here, figure 2 shows the data from ANSI S3.4 [1], with black dots indicating the tabulated values [1, tab. 6] and a black contour indicating the data presented graphically [1, fig. 8].

The comparison of the tabulated values and the graph indicates that the tabulated values are somewhat upward shifted compared to the graph. The reason is presumably that figure 8 of ANSI S3.4 [1] appears to be erroneously taken from [6, fig. 7], where the normalized high-frequency absolute-threshold excitation $E'_{\mathrm{THQ,500\,Hz}}$ is 2.31 [6, sec. 1.6], somewhat different from 2.36 of the standard [1, sec. 3.6.2]. This deviation affects $A(f_m)$ at frequencies f_m above 500 Hz, since $A(f_m \geq 500\,\mathrm{Hz})$ is defined as $2\,E'_{\mathrm{THQ,500\,Hz}}$ in both cases ([6, sec. 1.6] and [1, sec. 3.6.4]). Consequently, at $f_m \geq 500\,\mathrm{Hz}$ and therefore at $L_\mathrm{G}(f_m) = 0\,\mathrm{dB}$, the rightmost value of figures 8 of [1] and 2 of the paper on hand, the standard requires $A(f_m)|_{L_\mathrm{G}(f_m)=0\,\mathrm{dB}} = 4.72$, whereas [6] uses the assumption $A(f_m)|_{L_\mathrm{G}(f_m)=0\,\mathrm{dB}} = 4.62$. The black contour in figure 2 takes on the value $A(f_m)|_{L_\mathrm{G}(f_m)=0\,\mathrm{dB}} = 4.62$, whereas the tabulated data contain $A(f_m)|_{L_\mathrm{G}(f_m)=0\,\mathrm{dB}} \approx 4.72$. These values support our initial hypothesis that table 6 of [1] represents the data actually in

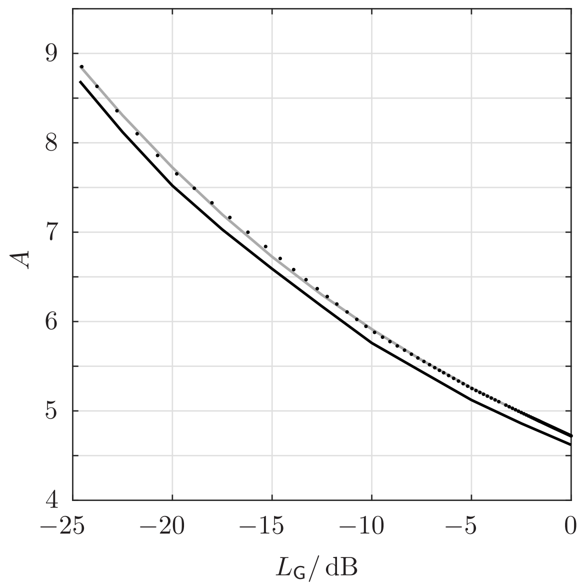

Figure 2: *Parameter $A(f_m)$ of ANSI S3.4-2007 [1] as a function of the so-called "low-level gain of the cochlear amplifier" $L_G(f_m)$. Black dots indicate the tabulated data [1, tab. 6], whereas the black contour represents the data presented graphically [1, fig. 8]. The gray contour was calculated according to equation 8 proposed in this paper.*

line with the other parameters of ANSI S3.4-2007 [1], whereas figure 8 of [1] was taken from [6] and fails to comply with the rest of the standard.

Figure 2 shows that table 6 of [1] contains an irregularity in the sampling in the range between $-3\,\mathrm{dB}$ and $-4\,\mathrm{dB}$: the samples at $L_\mathrm{G}(f_m) = -3.63\,\mathrm{dB}$ and $L_\mathrm{G}(f_m) = -3.27\,\mathrm{dB}$ are separated by almost $0.4\,\mathrm{dB}$ whereas the neighboring samples are only separated by $0.2\,\mathrm{dB}$ or less (cf. black dots in figure 2 in the range of $L_\mathrm{G}(f_m) \approx -3\,\mathrm{dB}$). The reason for this non-monotonic sampling is unclear and may be corrected in a revised version.

An alternative to a revision of the table is replacing the table by an equation. In line with the formulae proposed above, $A(f_m)$ tabulated in [1] can be approximated by

$$A(f_m) = 2.4464 + 2.2794\,G(f_m)^{-0.1823} = 2.4464 + 2.2794\,\left(\frac{E'_{\mathrm{THQ,500\,Hz}}}{E'_{\mathrm{THQ}}(f_m)}\right)^{-0.1823}. \quad (8)$$

This equation represents the tabulated $A(f_m)$ as a function of $L_\mathrm{G}(f_m)$ [1, tab. 6] on average within ± 0.0081, and with a maximum deviation of 0.0567. These deviations correspond to less than 0.2% and 0.9%, respectively (cf. gray contour in figure 2), which is considered acceptable for the purpose of loudness prediction (cf. section 5). While approximating the current data, the formula provides a unique, exponentially decaying definition of $A(f_m)$ as a function of $G(f_m)$ in line with the formulae proposed above. Thus, the second term of equation 8 is recommended for future use in a potential revision of the standard.

4.5 Loudness summation

The final step of predicting loudness according to Zwicker's proposal is integrating the specific-loudness pattern $N'(f_m)$ along f_m, that is across the spectral channels [2]. This is realized in the ANSI model by summing the specific loudnesses of the channels weighted by 0.1, as the critical-band-rate scale is defined in steps of 0.1 $\mathrm{ERB_N}$-numbers [1, sec. 3.7].

Due to the somewhat different structure, the ANSI model contains a binaural-summation step not present in Zwicker's procedure, which was developed for diotic sound presentation directly. This final step of ANSI S3.4 [1, sec. 3.9] results in the calculated loudness N.

5 INITIAL VERIFICATION

Providing an initial verification of the proposed modifications, the loudness N for selected examples from [1, A.1] was calculated with different algorithms. Pure tones with low and medium frequencies f in the audible range at different levels L were selected in order to cover a wide parameter range. Table 1 shows the (informative) sample results given by the standard (row [1, A.1]), the values predicted with the software accompanying ANSI S3.4 (row [1, SW]) and the results calculated using the software [17] in two different modes: a) according to the current ANSI, b) ANSI method with the modifications proposed here.

Table 1: Predicted loudness N of stationary pure tones (level L, frequency f, frontally-incident plane wave perceived binaurally in the free sound field). Tabulated data of ANSI S3.4 [1, A.1], predictions with the software accompanying the standard [1, SW], and with the software [17] in two modes: a) according to ANSI S3.4-2007, b) according to ANSI S3.4-2007 modified as proposed here.

L/ dB SPL	10	20	30	40	50	60	70	80	20	40	60	80	50
f/ kHz	1								3				0.1
N/ sone [1, A.1]	0.03	0.14	0.42	1.0	2.1	4.2	8.1	16.0	0.35	1.8	7.1	27.5	0.345
N/ sone [1, SW]	0.029	0.142	0.422	0.997	2.1	4.17	8.1	15.98	0.348	1.82	7.09	27.49	0.345
N/ sone [17, a)]	0.029	0.142	0.422	0.997	2.098	4.166	8.102	15.981	0.348	1.819	7.094	27.489	0.345
N/ sone [17, b)]	0.029	0.142	0.422	0.997	2.098	4.166	8.102	15.980	0.348	1.819	7.093	27.488	0.348

The predictions with the modified model (last row of table 1) are considered reasonably accurate compared to the reference data and the other methods. Also the different algorithms appear to agree well. In order to ensure standard conformity or to determine the amount of deviation, the standard requires rounding rules and normative sample results, with margins.

6 CONCLUSIONS

This contribution revisits the parameter specifications and procedure definition of the current American national standard ANSI S3.4-2007 [1], which describes a procedure for computationally predicting the loudness of steady sounds. Without addressing the algorithm's suitability for the purpose or the results' agreement with perceptual data, parameters are discussed regarding their consistency throughout the standard, mutual interrelations, and especially their actual specification in the document. The discussion shows that with regard to all the above-mentioned criteria, formulae and clearly specified variables could improve the current standard. A self-consistent set of such formulae is proposed, which provided a good accuracy in the initial validation of the loudness predictions. This study may contribute to a future revision of ANSI S3.4 [1] with a more thorough, unambiguous description of the procedure.

7 ACKNOWLEDGMENTS

The first author is indebted to Prof. Dr.-Ing. Werner Hemmert for his continuous support as well as for valuable discussions and comments.

8 REFERENCES

1. ANSI S3.4. Procedure for the Computation of Loudness of Steady Sounds. American National Standard, American National Standards Institute, Washington, 2007.
2. Eberhard Zwicker. Über psychologische und methodische Grundlagen der Lautheit (On psychological and methodical basics of loudness). *Acustica*, 8:237–258, 1958.
3. Eberhard Zwicker and Bertram Scharf. A model of loudness summation. *Psychological Review*, 72(1):3–26, 1965.
4. Eberhard Zwicker, Hugo Fastl, and C. Dallmayr. Basic-program for calculating the loudness of sounds from their 1/3-oct band spectra according to ISO 532 B. *Acustica*, 55: 63–67, 1984.
5. Hugo Fastl and Eberhard Zwicker. *Psychoacoustics – Facts and Models*. Springer, Berlin, Heidelberg, 3rd Edition, 2007.
6. Brian C. J. Moore, Brian R. Glasberg, and Thomas Baer. A model for the prediction of thresholds, loudness, and partial loudness. *J. Audio Eng. Soc.*, 45(4):224–240, 1997.
7. Brian R. Glasberg and Brian C. J. Moore. Prediction of absolute thresholds and equal-loudness contours using a modified loudness model. *J. Acoust. Soc. Am.*, 120(2):585–588, 2006.
8. DIN 45 631. Berechnung des Lautstärkepegels und der Lautheit aus dem Geräuschspektrum. Verfahren nach E. Zwicker (Calculation of loudness level and loudness from the sound spectrum. Zwicker method). Deutsche Norm, Beuth Verlag, Berlin, 1991.
9. ISO 532. Acoustics – Method for calculating loudness level. International Standard, International Organization for Standardization, ISO, Geneva, 1975.
10. DIN 45 631/A1. Berechnung des Lautstärkepegels und der Lautheit aus dem Geräuschspektrum – Verfahren nach E. Zwicker – Änderung 1: Berechnung der Lautheit zeitvarianter Geräusche; mit CD-ROM (Calculation of loudness level and loudness from the sound spectrum – Zwicker method – Amendment 1: Calculation of the loudness of time-variant sound; with CD-ROM). Deutsche Norm, Beuth Verlag, Berlin, 2010.
11. Hugo Fastl, Florian Völk, and Martin Straubinger. Standards for calculating loudness of stationary or time-varying sounds. In *Inter-Noise 2009*, 2009.
12. Jan Rennies, Jesko L. Verhey, and Hugo Fastl. Comparison of loudness models for time-varying sounds. *Acta Acustica united with Acustica*, 96:383–396, 2010.
13. Florian Völk. Loudness and noise rating: How may standardization bridge the gap between science and engineering? *POMA*, 25(1):http://dx.doi.org/10.1121/2.0000137.
14. Florian Völk. Comparison and fitting of analytical expressions to existing data for the critical-band concept. *Acta Acustica united with Acustica*, 101(6):1157–1167, 2015.
15. Klaus Betke. New hearing threshold measurements for pure tones under free-field listening conditions. *J. Acoust. Soc. Am.*, 89(5):2400–2403, 1991.
16. DIN EN ISO 389-7. Akustik – Standard-Bezugspegel für die Kalibrierung von audiometrischen Geräten – Teil 7: Bezugshörschwellen unter Freifeld- und Diffusfeldbedingungen (Acoustics – Reference zero for the calibration of audiometric equipment – Part 7: Reference threshold of hearing under free-field and diffuse-field listening conditions). Deutsche Norm, Beuth Verlag, Berlin, 2006.
17. WindAcoustics Suite. Modular software package for auditory-adapted signal and system analysis, real-time audio-signal processing, audio recording, and sound measurements. WindAcoustics UG [haftungsbeschränkt], www.windacoustics.com, 2016.