Loudness and Sharpness Calculation

Psychoacoustics is the science of the relationship between physical quantities of sound and subjective hearing impressions. To examine these relationships, physical parameters, such as sound pressure level, frequency and modulation depth, are mapped to hearing-related parameters. Unlike the physical quantities, these hearing-related quantities – also referred to as psychoacoustic parameters – provide a linear representation of human hearing perception. This means that a doubling of a psychoacoustic quantity corresponds to a doubling of the corresponding subjective perception level.
ArtemiS SUITE offers the possibility to calculate various psychoacoustic parameters. This Application Note explains how the psychoacoustic quantities loudness and sharpness can be calculated and used in ArtemiS SUITE.

The psychoacoustic parameters roughness, fluctuation strength and tonality have been described in the Application Note “Psychoacoustic Analyses II”, which you can download in the Download Center of our web site.

Technical terms used

Critical bands

Various experiments and hearing tests have shown that human hearing combines sound stimuli which are situated in close proximity of each other in frequency into particular frequency bands. These bands are called “critical bands”. In serializing these frequency bands a frequency scale is created which is called “critical band rate” and which is measured in the unit “Bark”. The audible frequency range was arranged by Zwicker into 24 critical bands on a scale from 0 to 24 Bark ([1], [2]). The critical band width increases with increasing frequency. Table 1 (next page) displays this arrangement. Another distribution of the frequency groups is described by the ERB scale (Equivalent Rectangular Bandwidth). The width of the frequency groups on the ERB scale is different from the width of the frequency groups on the Bark scale.

Critical band rate z [Bark]	**Frequency f [Hz]**	**Δf [Hz]**	**Critical band rate z [Bark]**	**Frequency f [Hz]**	**Δf [Hz]**	**Critical band rate z [Bark]**	**Frequency f [Hz]**	**Δf [Hz]**
0	0		8	920		16	3150	
		100			160			550
1	100		9	1080		17	3700	
		100			190			700
2	200		10	1270		18	4400	
		100			210			900
3	300		11	1480		19	5300	
		100			240			1100
4	400		12	1720		20	6400	
		110			280			1300
5	510		13	2000		21	7700	
		120			320			1800
6	630		14	2320		22	9500	
		140			380			2500
7	770		15	2700		23	12000	
		150			450			3500
						24	15500	

Table 1: The relation between critical band rate z and frequency f of [1]

Loudness

Loudness is the sensation value of the human perception of sound volume. By means of this parameter the human sensation of sound volume of acoustic signals is visualized on a linear scale. The unit of loudness is “sone” (derived from sonare, from Latin: sound). A sine tone of the frequency 1 kHz with a level of 40 dB has by definition a loudness of 1 sone. The loudness scale is distinguished by the fact that a tone which is perceived to have double the loudness on the loudness scale is designated by a doubled sone value. The loudness of sine tones and complex sounds was determined in hearing tests through comparison of loudness versus a 1 kHz sine tone. The determination of loudness has been specified in different standards.

Sound pressure level

The sensation of sound volume of human hearing is dependent on frequency. Thus sound events of equal level but different frequency do not always evoke the same sensation of sound volume in human beings. The volume level in the unit “phon” designates the sound pressure level of a 1 kHz sine tone which produces the same sensation of sonic volume as the tested sound event. Example: A sine tone at the frequency of 500 Hz, which is perceived to be as loud as a 1 kHz sine sound of 50 dB is designated a sound pressure level of 50 phon. According to DIN 45631 the sound pressure level L_N of the loudness N can be calculated as follows:

$$L_N = \begin{cases} 40 + 33.22 \cdot \lg\left(\dfrac{N}{\text{sone}}\right) & \text{for } N \geq 1\,\text{sone} \\ 40 \cdot \left(\dfrac{N}{\text{sone}} + 0.0005\right)^{0.35} & \text{for } N < 1\,\text{sone} \end{cases}$$

Specific loudness

The specific loudness N′ exhibits the distribution of loudness across the critical bands. Its unit is “sone/Bark”. The total loudness N is the result of the specific loudnesses N′ through integration of the critical band rate:

$$N = \int_0^{24\text{Bark}} N'(z)\,dz$$

Sharpness

The sharpness is a sensation value which is caused by high frequency components in a given noise. The unit of sharpness is “acum” (derived from acum, from Latin: sharp). Sharpness delineates human sensation in a linear manner as well. The value of 1 acum is attributed to a narrow-band noise at 1 kHz with a bandwidth smaller than 150 Hz and a level of 60 dB. The calculation of sharpness has been specified in the DIN 45692 standard.

Loudness calculation

ArtemiS SUITE offers several analyses for calculating the loudness, e.g. ***Loudness vs. Time*** / ***RPM*** and ***Specific Loudness vs. Time*** / ***RPM***.

There are various methods to calculate loudness, each of which is described in its own standard. In ArtemiS SUITE, the following four calculation methods are available [1]:

- ***DIN 45631/A1***
- ***ISO 532-1***
- ***ANSI S3.4 2007 (FFT)***
- ***ANSI S3.4 2007 (FFT/3rd Octave)***

Loudness calculation according to DIN 45631/A1

DIN 45631 (1967) standardizes a graphic procedure according to Zwicker, through which a specific loudness pattern of stationary noise can be established first from third-octave levels and from there the loudness level and loudness. This procedure was specified in the DIN 45631 (1991) standard with a computer program and instructions for the correction of low frequency components according to the curves of equal loudness. For time-variant sound events, however, this method turned out to deliver loudness values that were too low. Therefore, the DIN 45631/A1 standard was published in 2010. The A1 amendment extends DIN 45631 with a method to determine the loudness of time-variant sound.

To calculate loudness according to this standard, first a third-octave spectrum is determined using a digital filter bank. From these third-octave levels, then the main and accessory loudness values required to determine the specific loudness pattern are calculated using the calculation program specified in the standard. The total loudness is then determined by integrating the specific loudness pattern. You can find a detailed description of the calculation rules in the annex to this Application Note.

Since the human ear has a directional response pattern and therefore sound coming in from different directions is perceived with different sensitivities, the standard differentiates between loudness calculations for free fields and diffuse fields. The underlying calculation model of the standard delivers loudness values for the free field. In addition, the standard specifies level correction values for determining loudness in a diffuse field. According to the selected sound field, the unit *sone* is marked with an index *F* (for free field) or *D* (for diffuse field). Furthermore, the unit is marked with the index *G*, since the calculation is based on frequency groups.

A special feature of the ***Loudness vs. Time*** analysis used with ***DIN 45631/A1*** is that the default single value[2] calculated for this analysis is the N_5 value (the 5 % percentile value of the time-dependent loudness curve). This distinguishes this analysis from other ArtemiS SUITE analyses, where the single value in the diagram always represents the arithmetic average value of the curve by default.

[1] For the analyses ***Order Loudness vs. Time*** / ***RPM***, the Properties window does not allow you to select the loudness methods described in the following. Regarding this, please read the section “Order loudness calculation of order curves” in the annex to this Application Note.

[2] You can configure the calculation of the single-value results in ArtemiS SUITE in the Properties window of a 2D analysis (see section “Single values from 2D results” in the Help System).

Loudness calculation according to ISO 532-1

The ISO 532-1[3] standard is based on the calculation rule described in DIN 45631/A1. The most important change is a more detailed specification of the individual calculation steps. This change is supposed to avoid insecurity regarding the practical implementation. However, in most application cases, these changes will cause only minimal differences in the calculation results compared to the ***DIN 45631/A1*** setting.

The ISO 532-1 standard provides source code with a reference implementation, which covers in detail all steps of a time-dependent loudness calculation for a time-domain signal. This source code includes, for example, an automatic conversion to 48 kHz as well as the application of filters with fixed filter coefficients.

Furthermore, the standard provides loudness values for 24 test signals (including tolerances), allowing the user's own implementation to be tested using these test signals.

Loudness calculation according to ANSI S3.4-2007

The calculation according to ANSI S3.4-2007 emerges from a publication by Glasberg und Moore [4]. Unlike the method described above, which is based on a graphical procedure, the ANSI method is a computer-based procedure.

The loudness calculation with the method ***ANSI S3.4-2007 (FFT/3rd octave)*** is exactly equivalent to the procedure described in the ANSI standard, where a 3rd octave spectrum is prescribed as the input data for the loudness calculation. In ArtemiS SUITE, this 3rd octave spectrum is determined by means of an FFT analysis.

The loudness calculation with the method ***ANSI S3.4-2007 (FFT)*** calculates an FFT spectrum, too, but it does not subsume it into 3rd octave levels in the further course of the procedure, instead it uses the individual nodes of the FFT analysis for the loudness calculation. This method uses the loudness calculation algorithm described in the ANSI standard, but processes a larger amount of input data, as the individual nodes are not subsumed. Due to the larger input data set, this method delivers more precise results.[4]

To determine the loudness, both ANSI methods calculate the excitation patterns of the frequency groups on the ERB scale. From these excitation patterns, the specific loudness values are then calculated and added up to determine the total loudness.

The ANSI S3.4 2007 standard covers only the loudness calculation for constant time domain signals. If nevertheless the time-dependent loudness is determined by means of the loudness methods ***ANSI S3.4-2007 (FFT/3rd octave)*** *or* ***ANSI S3.4-2007 (FFT)***, ArtemiS SUITE calculates FFT spectra at first. After that the loudness results of each FFT window are successively entered in a diagram. The FFT length, i.e. the number of samples averaged, can be specified in the Properties window.

Using the analyses in ArtemiS SUITE

Depending on the loudness methods chosen, various further settings can be accessed in the Properties window. Figure 1 shows the Properties window of the analysis ***Loudness vs. Time***: on the left hand side when the ***DIN 45631/A1*** method is selected and on the right side when the ***ANSI S3.4 2007 (FFT)*** method is selected.

[3] The ISO 532-1 standard is still in development; a final version has not been published yet. The algorithm implemented in ArtemiS SUITE is based on the currently available draft of ISO 532-1.

[4] Using the ***ANSI S3.4 2007 (FFT)*** method, the loudness calculation of the reference tone (1 kHz, 40 dB) has a result of exactly 1 sone. If the ***ANSI S3.4 2007 (FFT/3rd octave)*** is used, the result is slightly higher (1.17 sone).

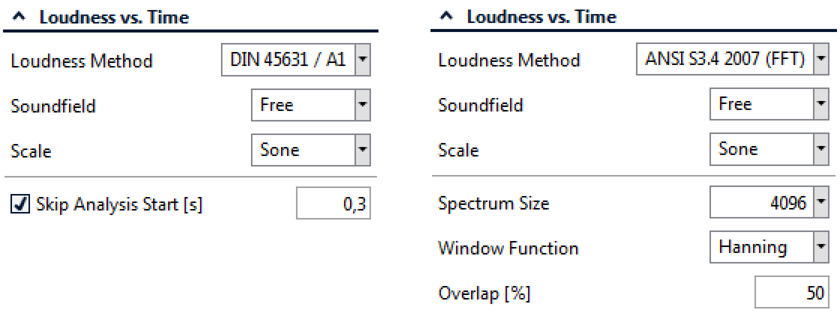

Figure 1: Properties window of the ***Loudness vs. Time*** analysis, left: ***DIN 45631/A1***, right: ***ANSI S3.4 2007 (FFT)***

The options ***Soundfield*** and ***Scale*** are available for both methods.

As described above, the sound pressure levels at the listener's ears and thus the perceived loudness of the sound event differ depending on the sound field. In order to account for these differences when calculating loudness, you can set the ***Soundfield*** select box for the loudness calculation either to ***Free*** or ***Diffuse*** field. Select the appropriate field type depending on your recording situation. If you made your measurements neither in a pure free field nor in a pure diffuse field, select the setting that fits your sound field better. Examples: For analyzing outdoor recordings, select ***Free*** even if the environment was not entirely non-reflective. For recordings made in a vehicle cabin, select ***Diffuse***, since this setting matches the conditions in such an environment more closely

The ***Scale*** setting defines the unit of the loudness. Depending on the setting, the results are displayed as loudness in ***sone*** or as loudness level in ***phon***.

The function ***Skip Analysis Start*** is only available for the DIN method. It allows the transient effect at the beginning of the analysis to be suppressed for the number of seconds specified. The transient effect is caused by the digital filters and can distort the total result. Enabling this function is particularly advisable if a single value calculation is to be performed.

If the ANSI method is selected, it is also possible to configure the DFT length (***Spectrum Size***) required for calculating the FFT, the ***Window Function*** and the ***Overlap***.

Examples

Figure 2 displays the results of the loudness analysis of two different time signals, calculated according to the loudness methods described above (blue: ***DIN 45631/A1***, magenta: ***ANSI S3.4 2007 (FFT)***, green: ***ANSI S3.4 2007 (FFT/3rd octave)***). For a better overview, the calculation results according to ***ISO 532-1*** have not been included. The curves would have been almost identical to the blue curves (***DIN 45631/A1***).

The left diagram shows the loudness curves for a pure sine tone. The blue curve (***DIN 45631/A1***) has the lowest values and the ***ANSI S3.4 2007 (FFT)*** curve (magenta) has the highest values.

The right diagram shows the loudness curves for a recording of broadband traffic noise in a big city. Again, the blue curve (***DIN 45631/A1***) has the lowest loudness values. However, the highest values are shown by the green ***ANSI S3.4 2007 (FFT/3rd Oct)*** curve this time. Furthermore, the distances between the curves in this diagram are greater than in the left diagram in many parts.

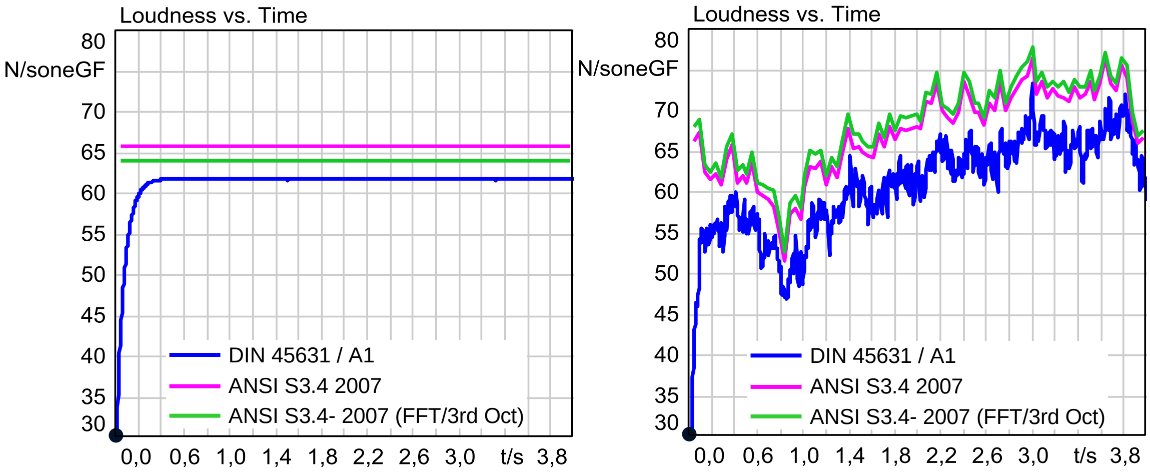

Figure 2: Comparison of the different loudness methods with two examples, blue: ***DIN 45631/A1***, magenta: ***ANSI S3.4 2007 (FFT)***, green: ***ANSI S3.4 2007 (FFT/3rd octave)***

Figure 3 shows another comparison. For this diagram, the loudness of a low-frequency combustion engine sound was calculated using the three methods. In this example, the highest loudness values are reached by the blue DIN 45631/A1 curve.

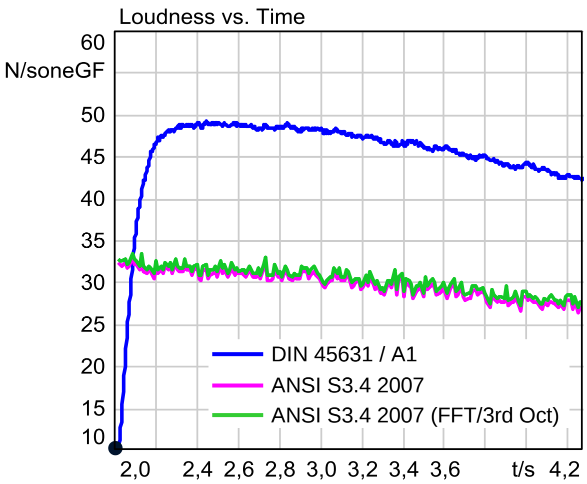

Figure 3: Comparison between the different loudness methods for a low-frequency sound event; blue: ***DIN 45631/A1***, magenta: ***ANSI S3.4 2007 (FFT)***, green: ***ANSI S3.4 2007 (FFT/3rd Oct)***

The comparison shows that the differences between the result curves depend on the type of sound, since the weighting of the frequency ranges differs between the methods. Generally, the ANSI method delivers higher sone values for broadband signals than the DIN method. For low-frequency signals, on the other hand, the calculation according to the DIN method delivers higher sone values.
The appropriate calculation method must therefore be chosen according to the type of sound to be examined and the objective of the examination.
Furthermore, the diagrams show that the loudness method ***DIN 45631/A1*** requires a certain settling time at the beginning of the signal, which is not required for the other methods. In the Properties window, it is therefore possible to exclude this transient effect by enabling the ***Skip Analysis Start*** option.

Sharpness Calculation

In ArtemiS SUITE you can calculate the sharpness by means of the analyses ***Sharpness vs. Time*** and ***Sharpness vs. RPM***.
Similar to loudness calculation, several methods are available for sharpness calculation, too:

- ***Aures***
- ***DIN 45692***
- ***von Bismarck***

In the Properties window of the analysis you can select the desired ***Sharpness Method*** (see figure 5).

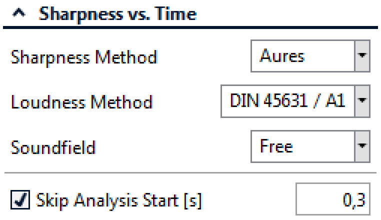

Figure 4: Properties window of the ***Sharpness vs Time*** analysis

Differences between the sharpness methods

The different sharpness methods differ in the following way: Von Bismarck developed a calculation procedure which is based on the distribution of the specific loudness throughout the critical band rate. To apply this method for calculating sharpness, select the ***von Bismarck*** setting.
The calculation method ***DIN 45692*** is based on research by Widmann [4] and is similar to the calculation method suggested by von Bismarck. Widmann performed his own listening tests and further adjusted the weighting functions for sharpness calculation determined by von Bismarck. Both methods were developed for calculating the sharpness of sound events with comparable loudness, which means that the influence of absolute loudness on the sharpness perception is not taken into account.
In contrast, Aures changed the calculation rules so that the influence of absolute loudness is accounted for, too. The ***Aures*** setting therefore allows you to calculate sharpness considering the absolute loudness of the signal.

Generally the sharpness calculation for all three methods is based upon the specific loudness distribution of the sound. In the Properties window of the sharpness analysis, you can therefore not only select the sharpness algorithm, but also the desired ***Loudness Method***. The available loudness methods and the corresponding additional parameter settings are the same as described above. If the ***DIN 45692*** setting is selected for the sharpness calculation, the ***Loudness Method*** is set to ***DIN 45631/A1*** automatically.

Due to the above-mentioned differences between the calculation methods, the results of a sharpness analysis can vary significantly. For that reason, when giving a sharpness factor, the calculation method should always be mentioned in order to avoid misunderstandings.

Examples

Figure 5 exemplifies the divergence between the three given calculation methods, by means of two noise samples (blue: ***Aures***, magenta: ***von Bismarck***, green: ***DIN 45692**, for all three curves, the loudness was calculated using the ***DIN 45631/A1*** method).
The difference between the calculation results of Aures and von Bismarck (or DIN 45692) is clearly visible, whereas the curves of von Bismarck and DIN 45692 show similar values. Furthermore, the figures show that the amount of difference depends on the type of sound. The difference is bigger for the first sound (pink noise) than for the second one (motorcycle). The artificial pink noise has been generated with a significantly higher level, therefore it has a higher loudness. For this reason, the value of the Aures sharpness is comparatively higher.

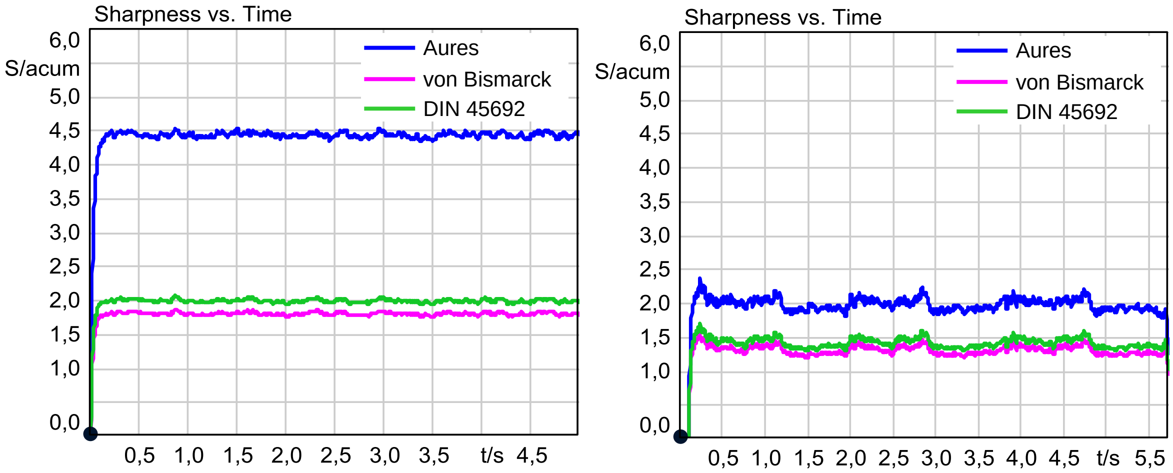

Figure 5: Comparison of the calculation methods regarding sharpness, blue: ***Aures***, magenta: ***von Bismarck***, green: ***DIN 45692***

The choice which may be the more appropriate sharpness method cannot be made on principle absolutely. The selection of the method has to be made according to the noises in question and the range of problems at issue. If you want to determine the sharpness of sound events with the same or similar loudness, the standardized ***DIN 45692*** method is a good choice.
Recent research has shown that in case of test signals with significant loudness differences, subjects take these loudness differences into account when assessing the sharpness of the sound. Therefore, the recommended setting for calculating the sharpness of such recordings is ***Aures***.

Notes

For calculating the analyses presented in this Application Note by means of a Pool Project, you need the following ArtemiS SUITE modules: **ASM 00** ArtemiS SUITE Basic Framework (code 5000), **ASM 01** ArtemiS SUITE Basic Analysis Module (code 5001), **ASM 12** ArtemiS SUITE Psychoacoustics Module (code 5012) and **ASM 13** ArtemiS SUITE Signature Analysis Module (Code 5013) for analyses against a reference quantity. If you want to calculate the analyses by means of an Automation Project or a Standardized Test Project, you may need other modules. Your HEAD acoustics representative will gladly provide you with further information.

Do you have any questions or comments?
Please write to imke.hauswirth@head-acoustics.de.
We look forward to receiving your feedback!

References

[1] Zwicker, E., „Psychoakustik“, Springer-Verlag, 1982
[2] Zwicker, E., „Unterteilung des hörbaren Frequenzbereichs in Frequenzgruppen“, Acustica 10, 185 (1960)
[3] Glasberg, B.R. und Moore, B.C., „Computation of Loudness of Steady Sounds”, J. Acoust. Soc. Am., Vol 120, 585-588, 2006Fastl, H. und Schmid, W., „Comparison of Loudness Analysis Systems”, Proceedings INTERNOISE 97 (Budapest, Ungarn), 25.-27. August, Seite 981-986
[4] Widmann, U. „Untersuchungen zur Schärfe und zur Lästigkeit von Rauschen unterschiedlicher Spektralverteilung“, DAGA '93, Bad Honnef, 1993, S. 644-647
[5] Aures, W., „Berechnungsverfahren für den Wohlklang beliebiger Schallsignale, ein Beitrag zur gehörbezogenen Schallanalyse“, Doktorarbeit, TU München, 1984
[6] Zwicker, E., „Dependence of post-masking on duration“, J. Acoust. Soc. Am., Vol 75, No. 1, Januar 1984

Annex

Order loudness calculation of order curves

When calculating the ***Order Loudness vs. Time / RPM*** analysis, it is not possible to select a loudness algorithm. These analyses use an algorithm developed by HEAD acoustics to calculate loudness. This algorithm is widely based on the method by W. Aures [5]. This loudness calculation method does not calculate 3rd octave levels first, but directly determines frequency group levels with a width of 1/5 Bark. The calculation is FFT-based and takes both the upper and the lower accessory loudness into account. Afterwards, the levels are corrected based on the transfer characteristics of the ear for a free field or a diffuse field.
Due to the higher frequency resolution, this algorithm is better suited, for example, for the analysis of engine run-ups.

Time-dependent loudness calculation according to DIN 45631/A1

Figure A.1 shows the calculation instruction for the time-dependent loudness according DIN 45631/A1. The individual components are explained as follows:

A) Calculation of third-octave levels in time
A filter bank with 28 Chebychev filters (low ripple) of the 6th order is used for the calculation.

B) Calculation of Intensity (Squaring)
In this phase of the processing the third-octave bands are established by squaring of time-dependent parameters of intensity.

C) Time-related averaging
The temporal succession is smoothed through lowpass filters.

D) Calculation of the main loudnesses
Calculation of the main loudnesses is according to the DIN norm standard. The signals of the lowpass filters 1-6, 7-9, as well as 10 and 11 are combined for the calculation. The signals of the lowpass filters 12-28 are processed individually.

E) Generation of a fade-out time depending on duration by means of a diode network
This effect is obtained when utilizing 4th order filtering by means of several lowpass filterings with varying time constants and a final maximum detection. A diode network described by Zwicker is utilized for the method with the filter of the 6th order [6].

F) Calculation of the loudness summation
Taking 20 main loudnesses, the specific loudness distribution is calculated initially. After that specific partial loudnesses are summed.

G) Temporal averaging of the loudness summation
The loudness summation is filtered with two lowpass filters of 1st order (time constant 3.5 and 70 ms). The following, weighted addition of these signals makes up the total loudness.

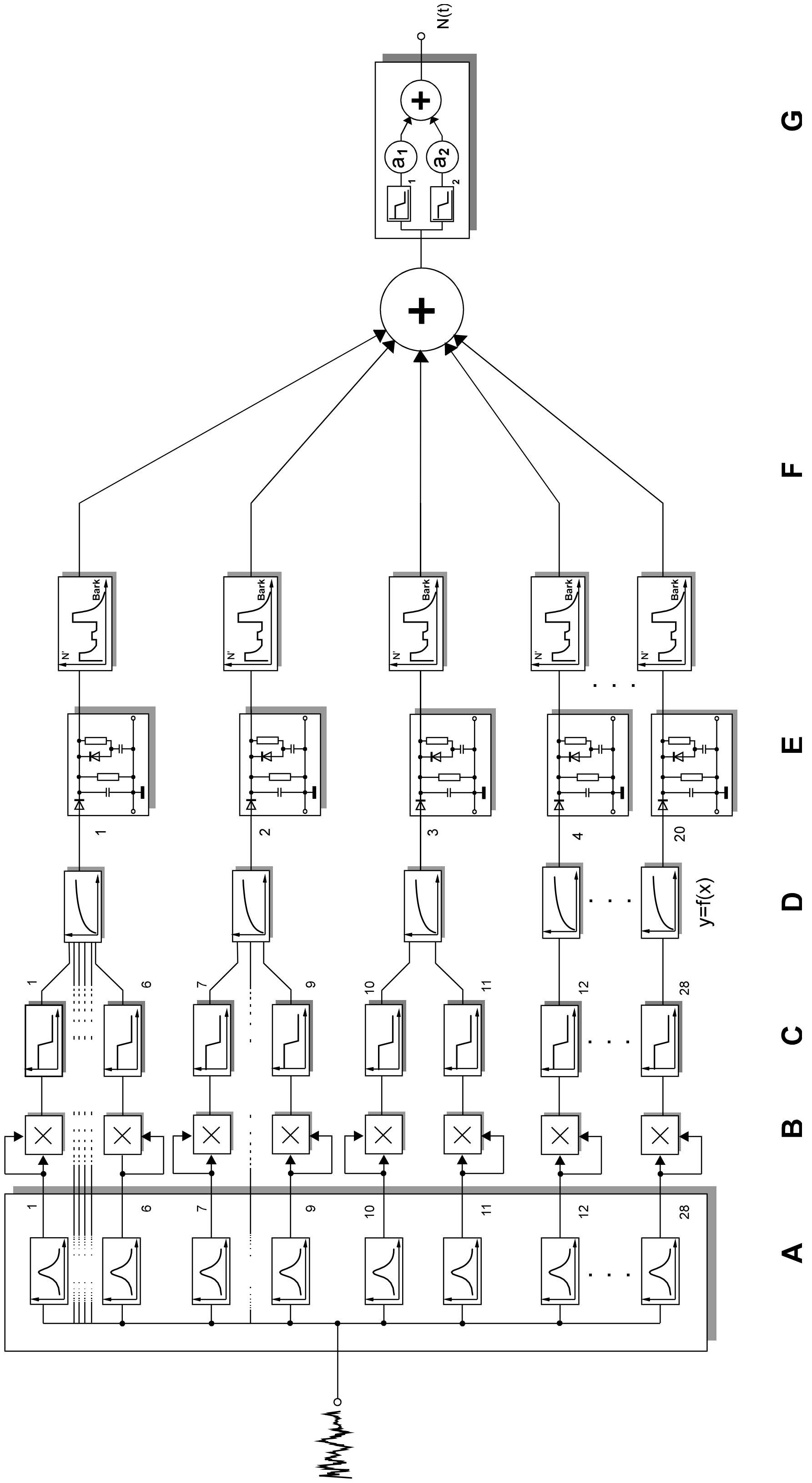

Figure A.1: Calculation of time-dependent loudness according to DIN 45631/A1