Loudness and Sharpness Calculation with ArtemiS

Introduction

The analysis software ArtemiS offers the possibility to calculate various psychoacoustic parameters. The calculation of those parameters is based on a processing of signals which has been closely modelled upon that of human perception. Thus these psychoacoustic parameters enable the user to analyse noise files with regard to the peculiarities of human hearing. In the following the calculation of the *loudness vs. time* analysis is described. In addition the *sharpness vs. time* analysis is presented which is based upon the analysis of loudness.

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 of 0 to 24 Bark ([1], [2]). The following table displays this arrangement:

Critical Band rate z [Bark]	f [Hz]	Δf [Hz]	Critical Band rate z [Bark]	f [Hz]	Δf [Hz]	Critical Band rate z [Bark]	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", lat.: 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 of stationary signals has been specified in the DIN 45631 and the ISO norm 532 B.

Sound Pressure Level

Tested tonal sounds of equal level but different frequency don't always evoke the same sensation of sound volume in human beings. The sensation of sound volume of human hearing is dependent on frequency. 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. That is to say 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:

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

Specific Loudness

The specific loudness 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" (of "acum", from the 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.

Loudness Calculation

ArtemiS makes a number of procedures for the analysis of loudness available. A choice of procedures and their respective settings can be found on the property page of the loudness analysis (see exhibit 1). In the first option of the menu ("calculation method") the user can choose one of six calculation methods: a) „DIN 45631 (/A1) " b) „FFT / ISO 532 B", c) „Filter / ISO 532 B" or

d) „FFT / HEAD“, e) „FFT / ANSI S3.4-2007“, f) „FFT(3rd oct) / ANSI S3.4-2007“. In accordance with the chosen method various further settings can be accessed. If the analysis based on FFT is chosen, the desired window function, spectrum size, *and overlap* can be set. The FFT / HEAD calculation enables the user furthermore to improve the resolution from 1 bark up to 1/5 bark. In using the calculation “Filter / ISO 532 B” various filter orders are available.

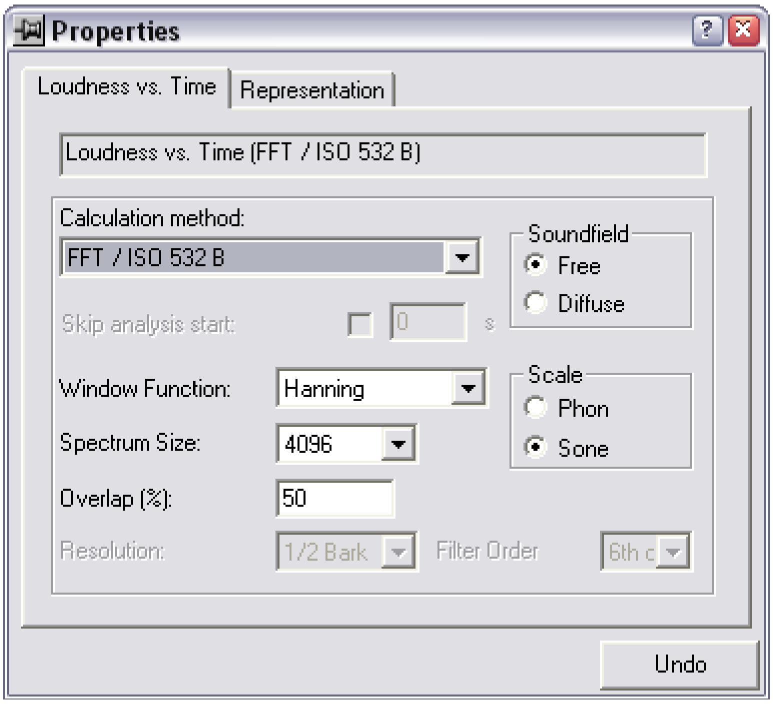

Figure 1: *Property Page* of the analysis of loudness with ArtemiS

Loudness Calculation according to ISO 532 B

The ISO-Norm standard 532 B standardizes a graphic procedure according to Zwicker, through which a specific loudness pattern can be established first from third-octave levels of stationary noise and from there the loudness level and loudness. This procedure is described as well in the German standard DIN 45631 and was specified in the DIN 45631 (1991) with a computer program and instructions for the correction of low frequency components according to the curves of equal loudness. The ArtemiS implementation is based upon this standard.
Furthermore, DIN 45631 indicates a BASIC program which approximates the said graphic procedure. The calculation of loudness in ArtemiS uses this BASIC program. The calculation methods “FFT / ISO 532 B” or “Filter / ISO 532 B” can be used in ArtemiS depending upon whether the third octave levels that are needed for the calculation shall be established on the basis of an FFT or through a filter bank. Figure 2 displays the calculation procedure in a schematically-reduced way. Both methods concur with the specifications given in the ISO-norm standard. The setting for the FFT windows (function, length, & overlap) or the order of the filters can be determined according to the choice made on the property page of the loudness-analysis.

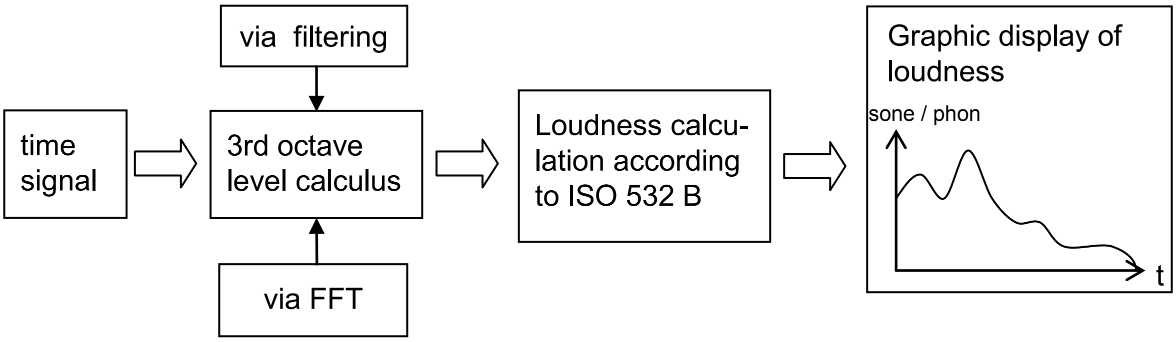

Figure 2: Procedure to determine loudness with ArtemiS in using the method FFT / ISO 532 or filter / 532

As the critical bands in the lower frequency range are broader than a third-octave band it is necessary to cluster the levels of frequencies under 250 Hz in several bands. For that purpose the given third octave levels are evaluated according to table which is specified in the norm standard and then added up by intensity. The critical bands of higher frequencies can be approximated by means of the third-octave bands in order to use the third octave level for further processing.
For the basic graphic procedure these levels are entered into a diagram. In addition to these levels, which are then marked up as main loudnesses, the so-called accessory loudnesses – as given by the norm standard – are entered into the diagram as well. The ensuing curve makes up the specific loudness pattern. The area below the curve constitutes the total loudness. As the ear possesses a spatial discriminating mechanism, and the sounds from different directions are perceived with a different level of loudness, the norm standard makes diagrams for the free field and the diffuse-field available. The unit sone has the index F (free field) or D (diffuse-field) according to the choice of the type of diagram. Furthermore the unit is designated with the index G, indicating that the calculation is carried out on the basis of the data of the critical bands.
Figure 3 displays such a configuration for the graphic evaluation of a noise in the free field. Within the diagram the partial loudnesses of a sample noise are marked up with the total loudness of 24 $sone_{GF}$. All lightly drawn lines belong to the standardized template. The partial loudnesses of the noise are entered as thick continuous lines. The area under the curve corresponds to the one under the dotted, horizontal line. The total loudness can be taken through the height of this line from the scales on the left and right (here: 24 $sone_{GF}$ or 86 $phon_{GF}$).

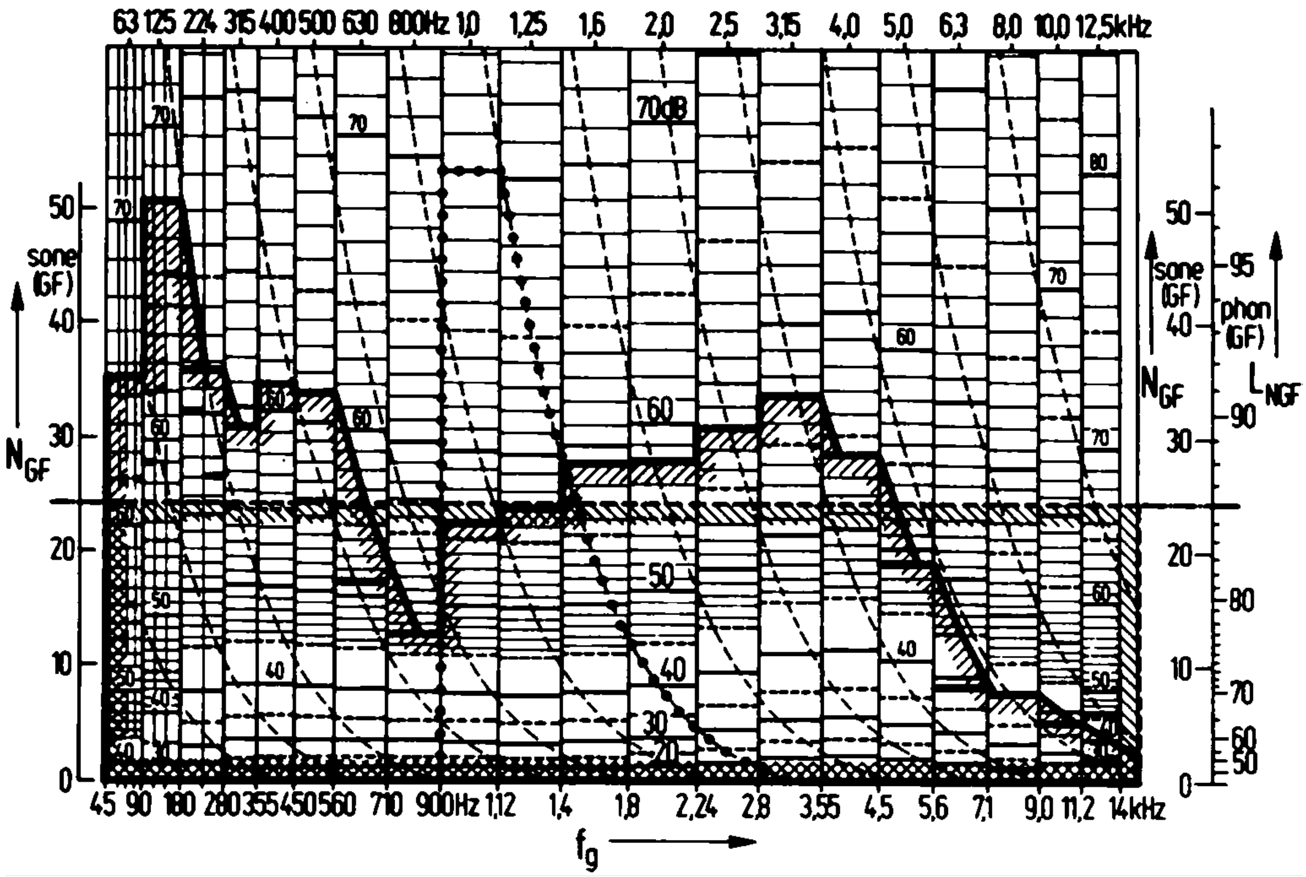

Figure 3: Application of the graphic procedure to calculate loudness N_{GF} = 24 sone (GF) and the loudness level L_{NGF} = 86 phon (GF) of a noise with the sound level of 73 dB. Partial loudnesses correspond partial areas / surfaces, the total loudness corresponds with the total area; dotted: partial loudnesses of a sinusoidal sound of kHz and 70 dB; from (1)

Loudness Calculation According to ANSI S3.4-2007

Besides the loudness calculation according to ISO 532B another algorithm, which is specified in the American ANSI S3.4-2007 standard, could be selected. This method emerges from a publication by Glasberg und Moore [6]. Unlike the method described above, which is based on a graphical procedure, the ANSI S3.4-2007 method is a computer-based procedure.

The "FFT(3rd octave) / ANSI S3.4-2007" method is exactly equivalent to a procedure described in the ANSI standard, where a 3rd octave spectrum is prescribed as the input data for the loudness calculation. In ArtemiS, this 3rd octave spectrum is determined by means of an FFT analysis. The „FFT / ANSI S3.4-2007 " method 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. [1]

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

To determine the loudness, both ANSI S3.4-2007 methods calculate the excitation patterns of the frequency groups on the ERB (Equivalent Rectangular Bandwidth)[2] scale. From these excitation patterns, the specific loudness values are then calculated and added up to determine the total loudness.
For broadband signals, the ANSI S3.4-2007 method delivers higher sone values than the ISO 532 B method. For low-frequency signals, on the other hand, the ISO 532 B method produces higher sone values.

Loudness Calculation According to DIN

This calculation method is identical to the method "Filter / ISO 532" described above, except that this method automatically uses 6th order filters. The calculation method "DIN 45631 / A1", that is available for time depending loudness analyses, complies with the new standard DIN 45631/A1 (see also "Loudness Calculation over Time").

Loudness Calculation on the basis of the HEAD Algorithm

This algorithm is determined largely by the procedure of W. Aures [3]. This method of loudness calculation does not determine third octave levels at the outset; instead the critical band levels are constituted directly. After this follows the correction of levels according to the transmission characteristics of the ear with regards to the free field or the diffuse field. Upon the basis of these levels the specific loudness can be calculated. The calculation instructions for this procedure vary slightly from the ones used for the calculation according to ISO 532 B. In this algorithm as well, the total loudness is calculated by integrating the specific loudnesses.
The HEAD algorithm makes it possible to increase the frequency resolution. In choosing the *resolution* setting on the property page the frequency resolution can be increased from 1 bark to 1/5 bark.

The Main Differences between the Loudness Algorithms "ISO 532 B" and "HEAD"

- The HEAD algorithm creates critical-bandwidth excitation levels, while the ISO procedure utilizes third octave levels in which several third octave levels are combined in the low frequency range, as the third-octaves are much narrower there than the critical bands.
- The HEAD algorithm adds the main excitation along with the accessory excitations, while the ISO procedure considers the maximum of the main excitation and the accessory excitations alone. Research about the summation of accessory masking proved that independent of various parameters there is an addition of the accessory excitations, which goes beyond the intensity addition and even beyond the amplitude addition.
- Low frequencies are under-represented by the ISO algorithm relative to the HEAD algorithm.
- In distinction to the HEAD algorithm, the ISO procedure does not take into account the accessory masking.
- The ISO procedure establishes the specific loudness pattern as a quasi-continuous or continuous function which is discovered through the critical band rate. The HEAD procedure on the other hand establishes the specific loudness pattern by means of discrete locations along the critical band scale whose distance can be adjusted from 1 to 1/5 bark.

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[2] The width of the frequency groups on the ERB scale is different from the width of the frequency groups on the Bark scale.

- The specific loudnesses of the excitation levels are calculated by means of slightly varying formulae.

Loudness Calculation over Time

The temporal effects of loudness are reproduced by means of filters in the methods "DIN 45631 / A1" and "Filter/ISO 532". They differ in the procedures applied with the 4th or 6th order filter settings.[3] In the annex to this Application Note, you will find a description of the different calculation procedures. In comparison, the calculation with 6th order filtering usually leads to smaller loudness values than the calculation with 4th order filters.
In the methods "FFT / ISO 532" and "FFT / HEAD", the levels from which the loudness is calculated are averaged over one FFT length, and the results are successively entered in a diagram. The FFT length, i.e. the number of samples averaged, can be specified in the settings dialog under "Window size".

The calculation of the time-dependent loudness is only standardized in the standard DIN 45631/A1. If the calculation method "DIN 45631 / A1" is selected in ArtemiS, the time-dependent loudness is calculated according to this standard. The calculation method "Filter / ISO 532 B" is compliant with this standard, too, as long as the 6th order filter setting is used. [4]
In addition to the calculation algorithms defined in this standard, further recommendations have been published [4]. These recommendations do not contain calculation algorithms, but target values and tolerance ranges for the stationary and time-dependent loudness of specific test signals. The calculation of loudness with the "DIN 45631 / A1" setting or the "Filter / ISO 532" setting with 6th order filtering is the best choice regarding the compliance with these recommendations.

Unlike the DIN 45631/A1 standard, which also describes the calculation of time-dependent loudness, the ISO 532 A/B standard covers only the loudness calculation for constant time domain signals.

A special feature of the analysis „loudness vs time" used with "DIN 45631 / A1 " is that the single value of this analysis being shown in the diagram is the N5 value (thus the 5% percentile value of the time-dependent loudness curve). This distinguishes this analysis from other ArtemiS analyses, where the single value in the diagram always represents the arithmetic average value of the curve.

Examples

Figure 4 displays the results of the loudness analysis of two different time signals, calculated according to the methods described above (green curve: "FFT / ISO 532 B", red curve: "Filter (6. Ordnung) / ISO 532 B", corresponds to "DIN 45631 / A1", turquoise curve: "FFT / HEAD algorithm"). The diagrams show, on the one hand, that the results vary markedly when different methods are chosen. The results of the different calculation methods deviate from each other

[3] This selection is only available for the "Filter / ISO 532 B" method. The "DIN 45631 / A1" method always uses 6th order filters.
[4] As already mentioned, the "DIN 45631/ A1" method uses the same algorithm as the calculation method "Filter / ISO 532" with the 6th order filter setting.

because of the different frequency distribution. Sound 1 is a purely sinusoidal sound. Sound 2 is decidedly a broadband noise. As the different calculation methods evaluate the frequency ranges in a different manner, different loudness diagrams are obtained.
On the other hand, it can be recognized that the calculation method "Filter / ISO 532" requires a certain amount of adjustment, which is not required for the other methods. In the property page, this transient effect can be hidden by entering a time value ≥ 0 s in the "Skip analysis start" field.

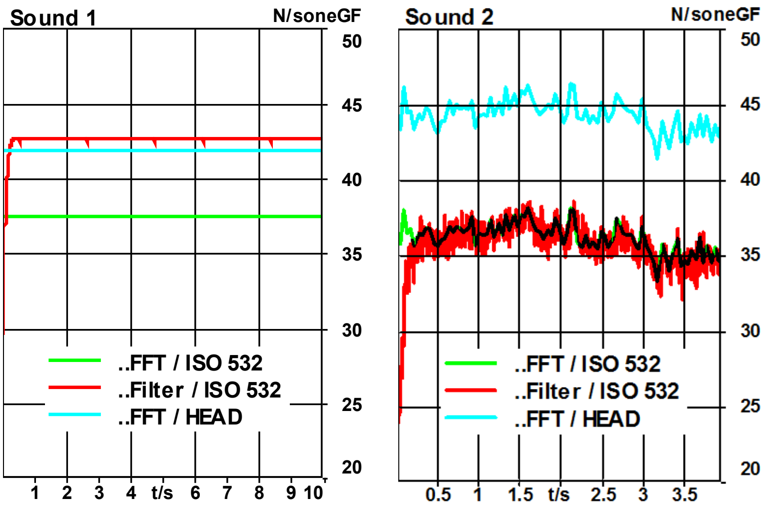

Figure 4: Comparison of the different calculation methods for loudness with two examples, green curve: "FFT/ISO 532 B", red curve: "Filter / ISO 532 B", turquoise curve: "FFT / HEAD algorithm"

For the selection of a suitable calculation algorithm, the type of sound to be examined and the purpose of the measurement must be considered. For example, an ISO calculation method must be chosen if the resulting loudness value is to comply with this standard. However, if the results of a listening test are to be modeled by means of psychoacoustic parameters, the HEAD algorithm can be an excellent choice in some cases.

Sharpness Calculation

The sharpness calculation is based upon the specific loudness distribution of the sound. Therefore the calculation of the loudness affects the results of the sharpness analysis. The loudness algorithms described above can be selected from the sharpness analysis property page in the field "loudness method" (re figure 5).
In addition to the various loudness algorithms, the ArtemiS user may choose from three algorithms to calculate sharpness: a) DIN 45692, b) Aures and c) von Bismarck.

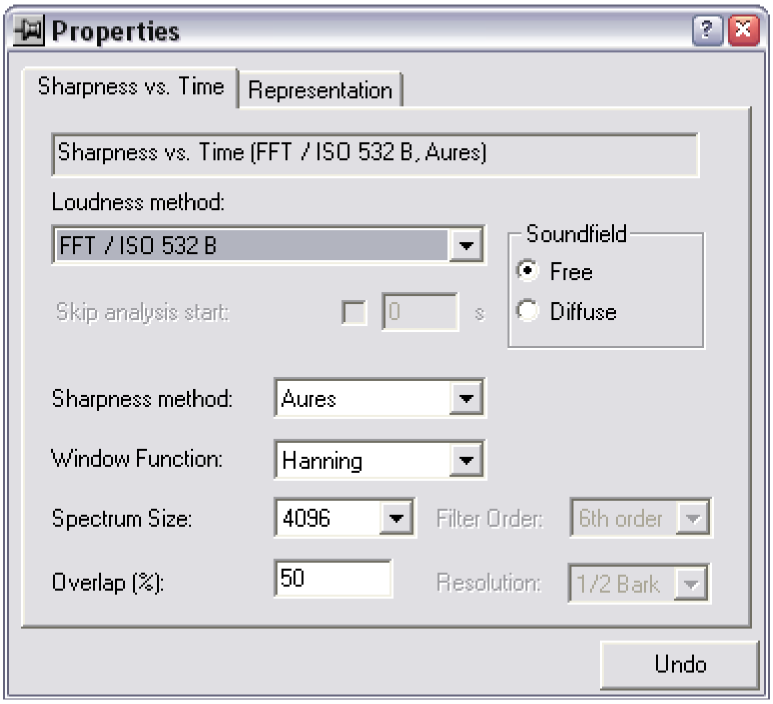

Figure 5: *Property page* of the sharpness analysis with ArtemiS

The results of the two 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. The procedure refers to sounds of equal loudness, meaning that the influence of the absolute loudness upon sharpness is not taken into consideration. Aures undertook a correction of the formula given by von Bismarck based on further results, changing the calculation instruction so that, that in addition, the influence of loudness is taken into account. The “DIN 45692” calculation method is similar to the method developed by von Bismarck. The “DIN 45692” method complies with the standard DIN 45692 that standardizes the sharpness calculation. If the “DIN 45692” setting is selected for the sharpness calculation, the method for the loudness calculation (Loudness method) is set to “DIN 45631 / A1” automatically.
The calculation methods discussed above vary considerably in their results. For that reason, when giving a sharpness factor, the calculation method should always be mentioned in order to avoid misunderstandings.
Figure 6 exemplifies the divergence between the two given calculation methods, by means of two noise samples (green curve: Aures Calculation, red curve: von Bismarck Calculation, turquoise curve: DIN 45692). The decisive difference between the two methods is clearly discernible. Furthermore the diagram demonstrates that the difference depends on the character of the noise under scrutiny. Sound 1 contains a white noise and sound 2 contains the noise of an electric motor.

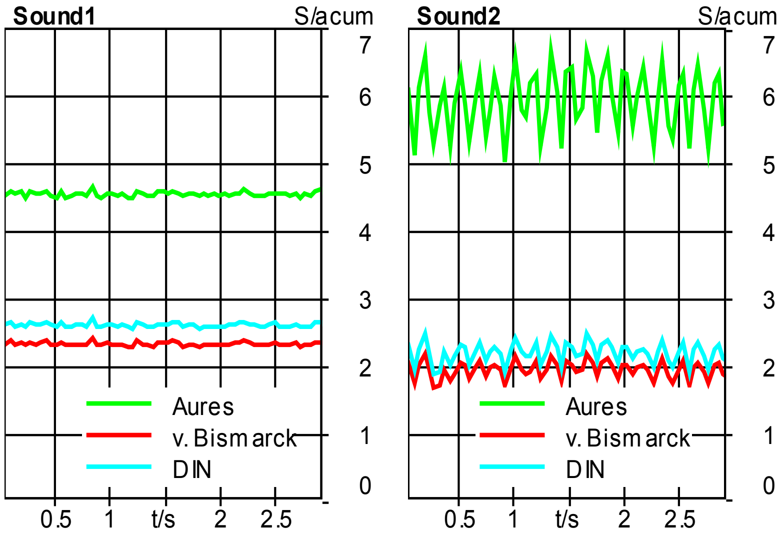

Figure 6: Comparison of the calculation methods regarding sharpness, green curve: Aures; red curve: von Bismarck, turquoise curve: DIN 45692

The choice which of the two may be the more appropriate 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. The scale of the sharpness values according to Aures is larger in comparison to the sharpness values of von Bismarck, viz. differences between the individual noises are put into greater relief. Thus this method can be advantageous, for example, for an application in which the noises should be categorized on the basis of the sharpness results according to the categories "working correctly" and "not working correctly" respectively, as the noises can be distinguished more easily because of greater variances of the sharpness values. On the other hand the case may occur that consideration of loudness according to the Aures method of sharpness calculation does not correspond to the subjective perception of tested individuals in the case of certain noise samples. In this case the calculation of sharpness following the method of von Bismarck or of DIN may yield a greater correlation between the evaluations of tested individuals and the results of the sharpness calculation.

Note

In order to use the features presented in this application note you will need the ArtemiS Basic version (Code 4600) and the ArtemiS Psychoacoustics Module, ATP 02 (Code 4602).

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

References

[1] Zwicker, E., „Psychoakustik", Springer Verlag 1982
[2] Zwicker, E., „Unterteilung des hörbaren Frequenzbereichs in Frequenzgruppen", Acustica 10, 185 (1960)
[3] Aures, W., „Berechnungsverfahren für den Wohlklang beliebiger Schallsignale, ein Beitrag zur gehörbezogenen Schallanalyse", dissertation, TU München, 1984

[4] Fastl, H. and Schmid, W., „Comparison of Loudness Analysis Systems", Proceedings INTERNOISE 97 (Budapest, Hungary), 25.-27. August, p. 981-986
[5] Zwicker, E., „Dependence of post-masking on duration", J. Acoustical Society of America, Volume 75, No. 1, January 1984

Annex

When calculating the time-related loudness with filters of the 4th order the output signals of the third-octave filter bank are processed initially through a lowpass filter. Then a specific loudness pattern is calculated from the results according to the ISO norm standard 532 B. In this process main loudnesses are weighted by means of a special filter, which has the effect that these main loudnesses increase rapidly with a given impulse and fade away with varying speed depending on the duration of the impulse.
Following this filtering the total loudness is determined by means of integrating the specific loudnesses. Afterwards the total loudness is put through a lowpass filter in order to reproduce the dependency of the loudness upon the lengths of the impulses. Impulses longer than approx. 100 – 200 ms evoke the maximum sensation of loudness. Shorter impulses decrease the sensation in such a manner that reducing the length of an impulse by a factor of 10 results in halving the loudness. This process is displayed schematically in figure A.1.
Figure A.2 shows the calculation instruction with filters of the 6th order. 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 ISO 532 B norm standard. The signals of the lowpass filters 1 – 6, 7 – 9, as well as 10 and 11 are combined for the calculation. Those 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 [5].
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.

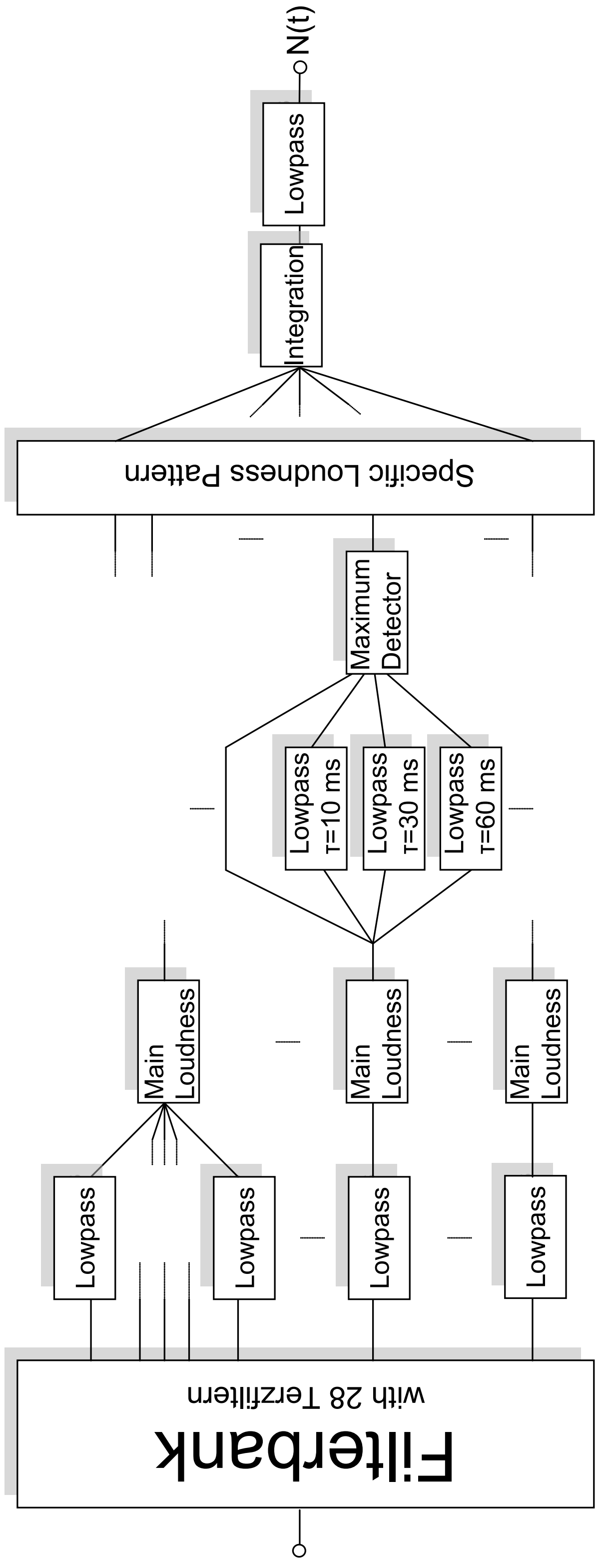

Figure A.1: Calculation of loudness on the basis of filters of the 4th order

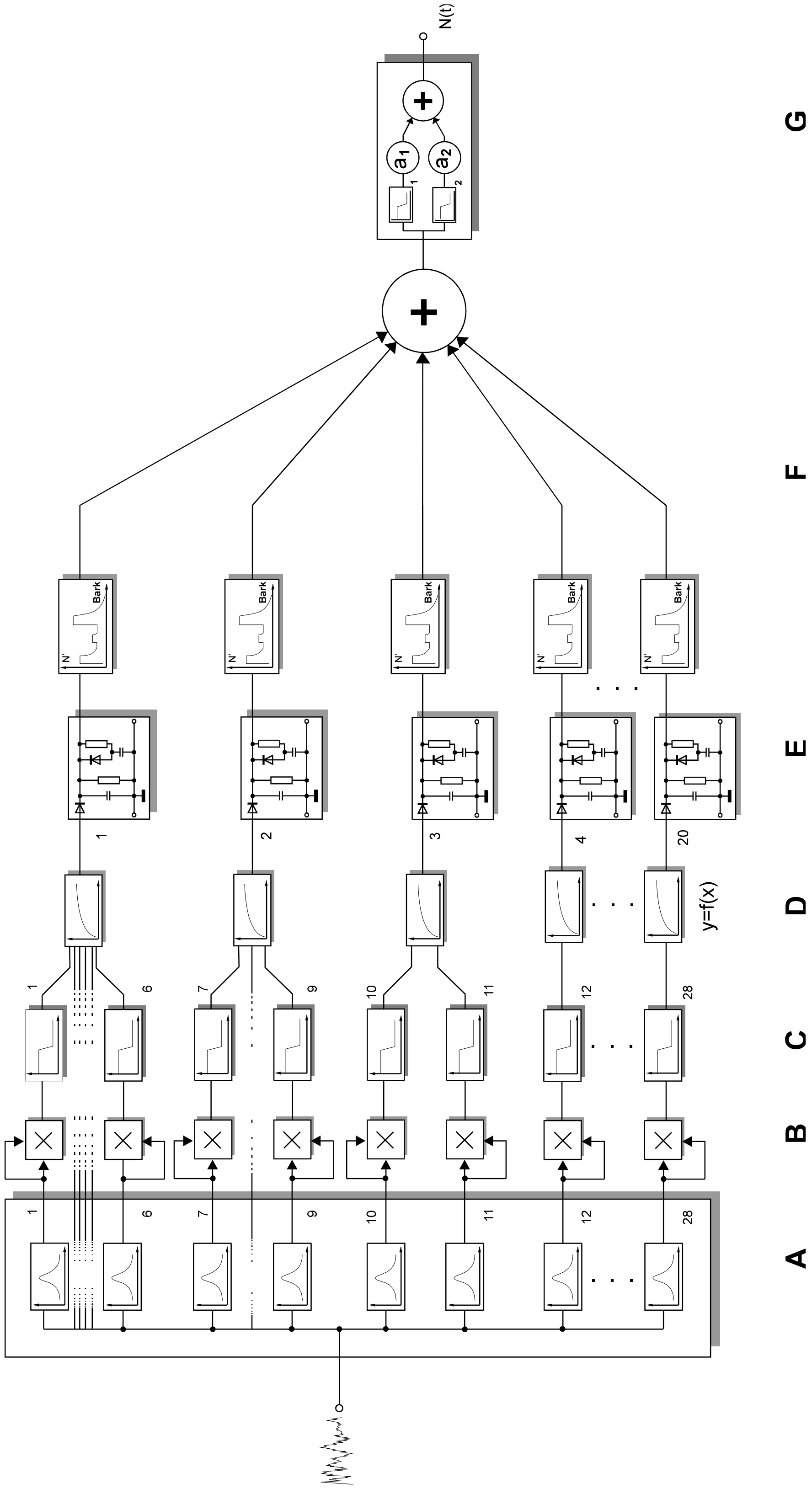

Figure A.2: Calculation of loudness on the basis of filters of the 6th order