1

Perception of sound

The perception of sound incidents requires the presence of some simple physical effects. A sound source oscillates and brings the surrounding air into motion. The compressability and mass of the air cause these oscillations to be transmitted to the listener's ear.

Small pressure fluctuations, referred to as sound pressure p, occur in air (or gas or fluid) and which are superimposed to the atmospheric pressure p_0. A spatially distributed sound field radiates from the source with different instantaneous sound pressures at each moment. The sound pressure is the most important quantity to describe sound fields and os always space- and time-dependent.

The observed sound incident at a point has two main distinguishing attributes: 'timbre' and 'loudness'. The physical quantity for loudness is sound pressure and the quantity for timbre is frequency f, measured in cycles per second, or Hertz (Hz). The frequency range of technical interest covers more than the range that is audible by the human ear, which is referred to as hearing level. The hearing range starts at about 16 Hz and ranges up to 16000 Hz (or 16 kHz). The infrasound, which is located below that frequency range, is less important for air-borne sound problems, but becomes relevant when dealing with vibrations of structures (e.g. in vibration control of machinery). Ultrasound begins above the audible frequency range. It is used in applications ranging from acoustic modelling techniques to medical diagnosis and non-destructive material testing.

The boundaries of the audible frequency range dealt with in this book cannot be defined precisely. The upper limit varies individually, depending on factors like age, and also in cases of extensive workplace noise exposure or the misuse of musical devices. The value of 16 kHz refers to a healthy, human being who is about 20 years old. With increasing age, the upper limit decreases by about 1 kHz per decade.

The lower limit is likewise not easy to define and corresponds to flickering. At very low frequencies a series of single sound incidents (e.g. a series of impulses) can be distinguished as well. If the frequency increases above the

flickering frequency of (about) 16 Hz, single incidents are no longer perceived individually, but seem to merge into a single noise. This transition can be found, for example, when it slowly starts to rain: the knocking of single rain drops at the windows can be heard until the noise at a certain density of rain merges into a continuous crackling. Note that the audible limit for the perception of flickering occurs at the same frequency at which a series of single images in a film start to appear as continuous motion.

The term 'frequency' in acoustics is bound to pure tones, meaning a sinusoidal wave form in the time-domain. Such a mathematically well-defined incident can only rarely be observed in natural sound incidents. Even the sound of a musical instrument contains several colourations: the superposition of several harmonic (pure) tones produces the typical sound of the instrument (see Fig. 1.1 for examples). An arbitrary wave form can generally be
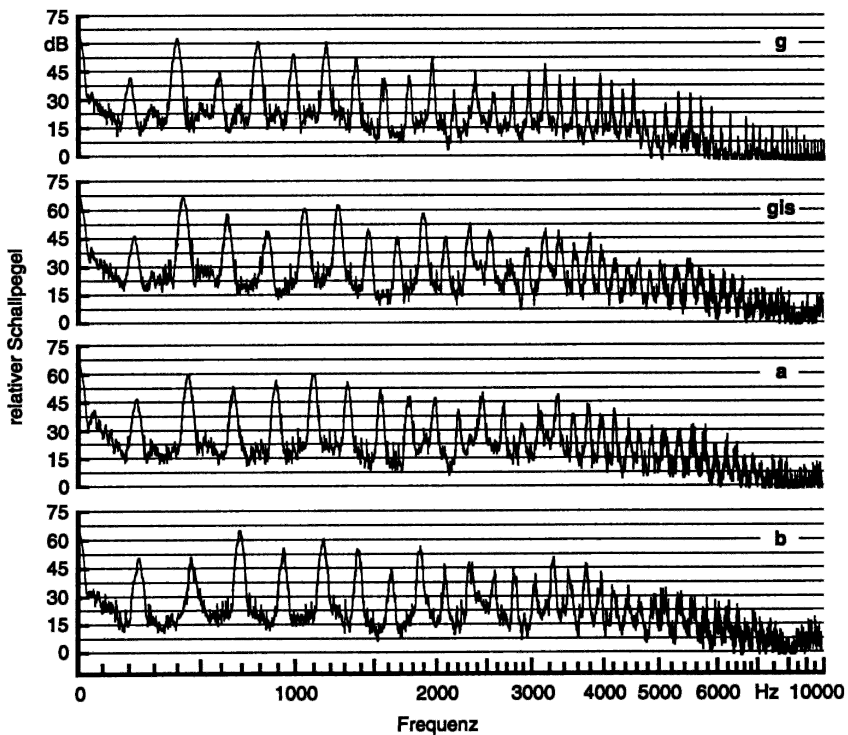
Fig. 1.1. Sound spectra of a violin played at different notes (from: Meyer, J.: "Akustik und musikalische Aufführungspraxis". Verlag Erwin Bochinsky, Frankfurt 1995). Relative sound pressure level versus frequency.

represented by its frequency components extracted through spectrum analy-

sis, similar to the analysis of light. Arbitrary signals can be represented by a sum of harmonics (with different amplitudes and frequencies). The association of decomposed time signals directly leads to the representation of the acoustic properties of transducers by their frequency response functions (as, for example, those of walls and ceilings in building acoustics, see Chap. 8). If, for instance, the frequency-dependent transmission loss of a wall is known, it is easy to imagine how it reacts to the transmission of certain sound incidents like, for example, speech. The transmission loss is nearly always bad at low frequencies and good at high frequencies: speech is therefore not only transmitted 'quieter' but also 'dull' through the wall. The more intuitive association that arbitrary signals can be represented by their harmonics will be sufficient throughout this book in the aforementioned hypothetical case. The expansion of a given signal into a series of harmonics, the so-called Fourier series and Fourier integrals, is based on a solid mathematical foundation of proof (see the last chapter of this book).

The subjective human impression of the sound pitch is perceived in such a way that a tonal difference of two pairs of tones is perceived equally if the ratio (and not the difference) of the two frequency pairs is equal. The tonal difference between the pair made of f_{a1} and f_{a2} and the pair made of f_{b1} and f_{b2} is perceived equally if the ratio

$$\frac{f_{a1}}{f_{a2}} = \frac{f_{b1}}{f_{b2}}$$

is valid. The transition from 100 Hz to 125 Hz and from 1000 Hz to 1250 Hz is, for example, perceived as an equal change in pitch. This law of 'relative tonal impression' is reflected in the subdivision of the scale into octaves (a doubling in frequency) and other intervals like second, third, fourth and fifth, etc. used for a long time in music. All of these stand for the ratio in frequency and not for the 'absolute increase in Hz'.

This law of 'tonal impression,' which more generally means that a stimulus R has to be increased by a certain percentage to be perceived as an equal change in perception, is not restricted to the tonal impression of the human being. It is true for other human senses as well. In 1834, Weber conducted experiments using weights in 1834 and found that the difference between two masses laid on the hand of a test subject was only perceived equally, when a mass of 14 g was increased by 1 g and a mass of 28 g was increased by 2 g. This experiment and the aforementioned tonal perception leads to the assumption that the increment of a perception ΔE for these and other physical stimuli is proportional to the ratio of the absolute increase of the stimulus ΔR and the stimulus R

$$\Delta E = k\frac{\Delta R}{R}, \qquad (1.1)$$

where k is a proportionality constant. For the perception of pitch the stimulus $R = f$ represents the frequency, for the perception of weight $R = m$ represents the mass on the hand.

This law of relative variation (1.1) is also true for the perception of loudness. If a test subject is repeatedly presented with sound incidents consisting of pairs of sound pressure incidents p and $2p$ and $5p$ and $10p$ respectively, the perceived difference in loudness should be equal. The perception of both pitch and loudness should at least roughly follow the law of relative variation (1.1).

As mentioned earlier (1.1), there is a relativity law which governs variations in stimulus ΔR and in perception ΔE. It is, of course, also interesting to examine the relation between R and E. Given that it is, at best, problematic, if not presumably impossible, to quantify perceptions, the principal characteristics of the $E(R)$ function should be clarified. These 'perception characteristics' are easily constructed from the variation law, if two points of stimulus R and perception E are chosen as shown in Fig. 1.2. A threshold stimulus R_0 is defined, at which the perception starts: stimuli $R < R_0$ below the threshold are not perceivable. A minimal stimulus is needed to achieve perception at all. The second point is chosen arbitrarily to be twice the threshold $R = 2R_0$ and the (arbitrary) perception E_0 is assigned. The further characteristics re-
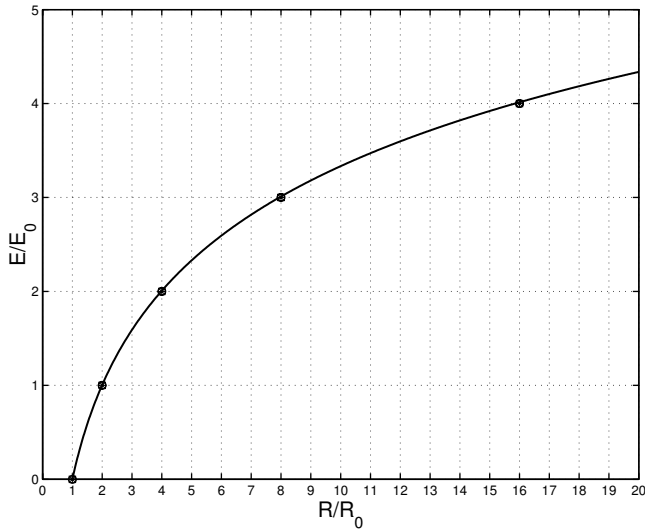
Fig. 1.2. Qualitative relation between stimulus R and perception E

sult from examining the perceptions $2E_0$, $3E_0$, $4E_0$, etc. The perception $2E_0$ is assigned twice the stimulus of E_0, therefore related to $R = 4R_0$. Just as $E = 3E_0$ is related to the stimulus $R = 8R_0$ the perception $4E_0$ is related to $R = 16R_0$, etc. As can be seen from Fig. 1.2 the gradient of the curve $E = E(R)$ decreases with increasing stimulus R. The greater the perception, the greater the increase of the stimulus has to be to achieve another increment of perception (for example E_0).

The functional relation $E = E(R)$ can certainly be determined from the variation law (1.1) by moving towards infinitesimal small variations $\mathrm{d}E$ and $\mathrm{d}R$:

$$\mathrm{d}E = k\frac{\mathrm{d}R}{R} \ .$$

Integration yields

$$E = 2.3k\lg(R/R_0). \tag{1.2}$$

Bare in mind that the logarithms of different bases are proportional, e.g. $\ln x = 2.3\lg x$. The perception of loudness is therefore proportional to the logarithm of the physical stimulus –in this case, sound pressure. This relation, validated at least roughly by numerous investigations, is also known as the Weber-Fechner-Law.

The sensual perception according to a logarithmic law (for the characteristics see Fig. 1.2 again) is a very sensible development of the 'human species'. Stimuli close to the threshold $R = R_0$ are emphasized and therefore 'well perceivable', whereas very large stimuli are highly attenuated in their perception; the logarithmic characteristics act as a sort of 'overload protection'. A wide range of physical values can thus be experienced (without pain) and several decades of physical orders of magnitude are covered. The history of the species shows that those perceptions necessary to survive in the given environment, which also cover a wide range of physical values, follow the Weber-Fechner-Law. This is not true for the comparatively smaller range of temperature perception. Variations of a tenth or a hundredth of a degree are by no means of interest to the individual. In contrast, the perception of light needs to cover several decades of order of magnitude. Surviving in the darkest night is as important as the ability to see in the sunlight of a very bright day. And the perception of weight covers a range starting from smallest masses of about 1 g up to loads of several 10000 g. The perception of loudness follows the logarithmic Weber-Fechner-Law, because the human ear is facing the problem of perceiving very quiet sounds, like the falling of the leaves in quiet surroundings, as well as very loud sounds, like the roaring sound of a waterfall in close vicinity. As a matter of fact, humans are able to perceive sound pressures in the range of $20 \cdot 10^{-6} N/m^2$ to approximately $200\,\mathrm{N/m^2}$, where the upper limit roughly depicts the pain threshold. About ten decades of loudness are covered, which represents an exceedingly large physical interval. To illustrate, this range in equivalent distances would cover an interval between 1 mm and 10 km. The amazing ear is able to perceive this range. Imagine the impossibility of an optical instrument (like a magnifying glass), to be able to operate in the millimeter range as well as in the kilometer range!

When technically quantifying sound pressure, it is more handy to use a logarithmic measure instead of the physical sound pressure itself to represent this wide range. The sound pressure level L is internationally defined as

$$L = 20\lg\left(\frac{p}{p_0}\right) = 10\lg\left(\frac{p}{p_0}\right)^2, \tag{1.3}$$

with $p_0 = 20\,10^{-6}\,\mathrm{N/m^2}$, as an expressive and easy to use measure. The reference value p_0 roughly corresponds to the hearing threshold (at a frequency of $1\,\mathrm{kHz}$, because the hearing threshold is frequency-dependent, as will be shown in the next section), so that $0\,\mathrm{dB}$ denotes the 'just perceivable' or 'just not perceivable' sound event. If not otherwise stated, the sound pressure p stands for the root mean square (rms-value) of the time domain signal. The specification in decibels (dB) is not related to a specific unit. It indicates the use of the logarithmic law. The factor 20 (or 10) in (1.3) is chosen in such a way that $1\,\mathrm{dB}$ corresponds to the difference threshold between two sound pressures: if two sound incidents differ by $1\,\mathrm{dB}$ they can just be perceived differently.

The physical sound pressure covering 7 decades is mapped to a $140\,\mathrm{dB}$ scale by assigning sound pressure levels, as can be seen in Table 1.1. Some examples for noise levels occurring in situations of every day life are also shown.

Table 1.1. Relationship between absolute sound pressure and sound pressure level

Sound pressure p (N/m², rms)	Sound pressure level L (dB)	Situation/description
$2\,10^{-5}$	0	hearing threshold
$2\,10^{-4}$	20	forest, slow winds
$2\,10^{-3}$	40	library
$2\,10^{-2}$	60	office
$2\,10^{-1}$	80	busy street
$2\,10^{0}$	100	pneumatic hammer, siren
$2\,10^{1}$	120	jet plane during take-off
$2\,10^{2}$	140	threshold of pain, hearing loss

It should be noted that sound pressures related to the highest sound pressure levels are still remarkably smaller than the static atmospheric pressure of about $10^5\,\mathrm{N/m^2}$. The rms value of the sound pressure at $140\,\mathrm{dB}$ is only $200\,\mathrm{N/m^2}$ and therefore $1/500$ of atmospheric pressure.

The big advantage when using sound pressure levels is that they roughly represent a measure of the perceived loudness. However, think twice when calculating with sound pressure levels and be careful in your calculations. For instance: How high is the total sound pressure level of several single sources with known sound pressure levels? The derivation of the summation of sound pressure levels (where the levels are in fact *not* summed) gives an answer to the question for incoherent sources (and can be found more detailed in Appendix A)

$$L_{\mathrm{tot}} = 10\lg\left(\sum_{i=1}^{N} 10^{L_i/10}\right), \qquad (1.4)$$

where N is the total number of incoherent sources with level L_i. Three vehicles, for example, with equal sound pressure levels produce a total sound pressure level

$$L_{\text{tot}} = 10 \lg \left(3\, 10^{L_i/10} \right) = 10 \lg 10^{L_i/10} + 10 \lg 3 = L_i + 4.8\,\text{dB}$$

which is 4.8 dB higher than the individual sound pressure level (and not three times higher than the individual sound pressure level).
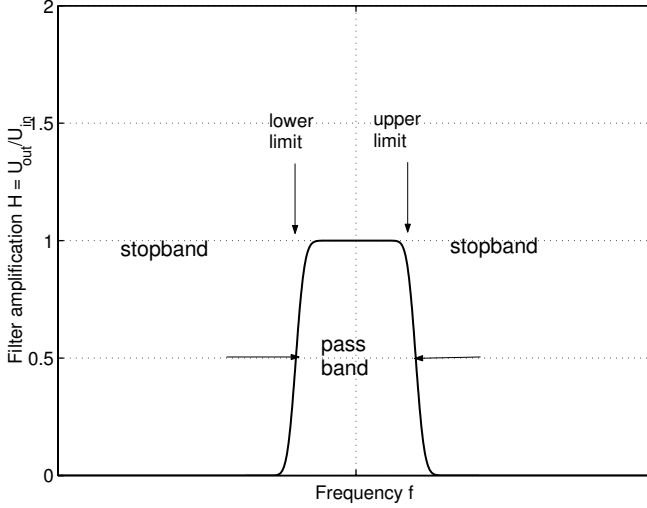
1.1 Octave and third-octave band filters

In some cases a high spectral resolution is needed to decompose time domain signals. This may be the case when determining, for example, the possibly narrow-banded resonance peaks of a resonator, where one is interested in the actual bandwidth of the peak (see Chap. 5.5). Such a high spectral resolution can, for example, be achieved by the commonly used FFT-Analysis (FFT: Fast Fourier Transform). The FFT is not dealt with here, the interested reader can find more details for example in the work of Oppenheim and Schafer "Digital Signal Processing" (Prentice Hall, Englewood Cliffs New Jersey 1975).

In most cases, a high spectral resolution is neither desired nor necessary. If, for example, an estimate of the spectral composition of vehicle or railway noise is needed, it is wise to subdivide the frequency range into a small number of coarse intervals. Larger intervals do not express the finer details. They contain a higher random error rate and cannot be reproduced very accurately. Using broader frequency bands ensures a good reproducibility (provided that, for example, the traffic conditions do not change). Broadband signals are also often used for measurement purposes. This is the case in measurements of room acoustics and building acoustics, which use (mainly white) noise as excitation signal. Spectral details are not only of no interest, they furthermore would divert the attention from the validity of the results.

Measurements of the spectral components of time domain signals are realized using filters. These filters are electronic circuits which let a supplied voltage pass only in a certain frequency band. The filter is characterized by its bandwidth Δf, the lower and upper limiting frequency f_l and f_u, respectively and the center frequency f_c (Fig. 1.3). The bandwidth is determined by the difference of f_u and f_l, $\Delta f = f_u - f_l$. Only filters with a constant relative bandwidth are used for acoustic purposes. The bandwidth is proportional to the center frequency of the filter. With increasing center frequency the bandwidth is also increasing. The most important representatives of filters with constant relative bandwidth are the octave and third-octave band filters. Their center frequency is determined by

$$f_c = \sqrt{f_l f_u}$$

The characteristic filter frequencies are known, if the ratio of the limiting frequencies f_l and f_u is given.

Fig. 1.3. Typical frequency response function of a filter (bandpass)

Octave bandwidth

$$f_u = 2f_l \ ,$$

which results in $f_c = \sqrt{2}f_l$ and $\Delta f = f_u - f_l = f_l = f_c/\sqrt{2}$.

Third-octave bandwidth

$$f_u = \sqrt[3]{2}f_l = 1.26f_l \ ,$$

which results in $f_c = \sqrt[6]{2}f_l = 1.12f_l$ and $\Delta f = 0.26f_l$.

The third-octave band filters are named that way, because three adjacent filters form an octave band filter ($\sqrt[3]{2}\sqrt[3]{2}\sqrt[3]{2} = 2$). The limiting frequencies are standardized in the international regulations EN 60651 and 60652.

When measuring sound levels one must state which filters were used during the measurement. The (coarser) octave band filters have a broader pass band than the (narrower) third-octave band filters which let contributions of a higher frequency range pass. Therefore octave band levels are always greater than third-octave band levels. The advantage of third-octave band measurements is the finer resolution (more data points in the same frequency range) of the spectrum.

By using the level summation (1.4), the octave band levels can be calculated using third-octave band measurements. In the same way, the levels of broader frequency bands may be calculated with the aid of level summation (1.4). The (non-weighted) linear level is often given. It contains all attributes of the frequency range between 16 Hz and 20000 Hz and can either be measured directly using an appropriate filter or determined by the level addition

of the third-octave or octave band levels in the frequency band (when converting from octave bands, $N = 11$ and the center frequencies of the filters are 16 Hz, 31.5 Hz, 63 Hz, 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz, 8 kHz and 16 kHz). The linear level is always higher than the individual levels, by which it is calculated.

1.2 Hearing levels

Results of acoustic measurements are also often specified using another single value called the 'A-weighted sound pressure level'. Some basic principles of the frequency dependence of the sensitivity of human hearing are now explained, as the measurement procedure for the A-weighted level is roughly based on this.
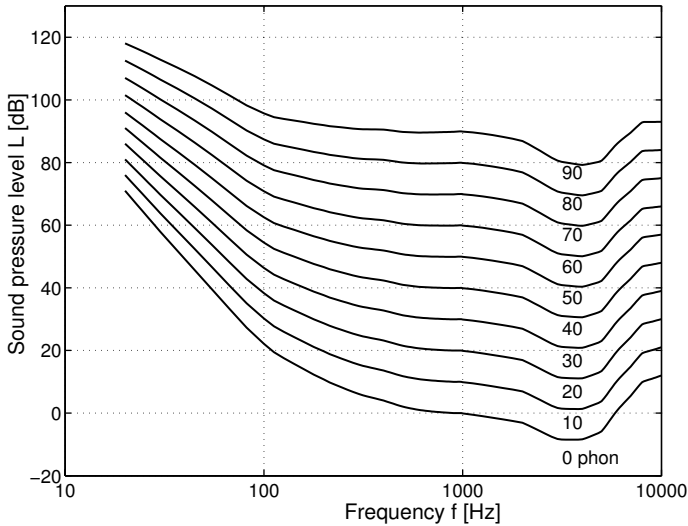

Fig. 1.4. Hearing levels

The sensitivity of the human ear is strongly dependent on the tonal pitch. The frequency dependence is depicted in Fig. 1.4. The figure is based on the findings from audiometric testing. The curves of perceived equal loudness (which have the unit 'phon') drawn in a sound pressure level versus frequency plot. One can imagine the development of these curves as follows: a test subject compares a 1 kHz tone of a certain level to a second tone of another frequency and has to adjust the level of the second tone in such a way that it is perceived with equal loudness. The curve of one hearing level is obtained by varying the frequency of the second tone and is simply defined by the level of the 1 kHz tone. The array of curves obtained by varying the

level of the 1 kHz tone is called hearing levels. It reveals, for example, that a 50 Hz tone with an actual sound pressure level of 80 dB is perceived with the same loudness as a 1 kHz tone with 60 dB. The ear is more sensitive in the middle frequency range than at very high or very low frequencies.

1.3 A-Weighting

The relationship between the objective quantity sound pressure or sound pressure level, respectively, and the subjective quantity loudness is in fact quite complicated, as can be seen in the hearing levels shown in Fig. 1.4. The frequency dependence of the human ear's sensitivity, for example, is also level-dependent. The curves with a higher level are significantly flatter than the curves with smaller levels. The subjective perception 'loudness' is not only depending on frequency, but also on the bandwidth of the sound incident. The development of measurement equipment accounting for all properties of the human ear could only be realized with a very large effort.
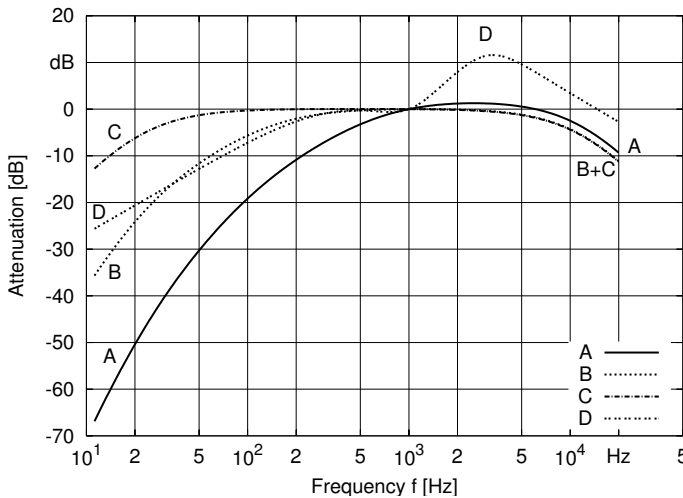
Fig. 1.5. Frequency response functions of A-, B-, C- and D-weighting filters

A frequency-weighted sound pressure level is used both nationally and internationally, which accounts for the basic aspects of the human ear's sensitivity and can be realized with reasonable effort. This so-called 'A-weighted sound pressure level' includes contributions of the whole audible frequency range. In practical applications the dB(A)-value is measured using the A-filter. The frequency response function of the A-filter is drawn in Fig. 1.5. The A-filter characteristics roughly represent the inverse of the hearing level

curve with 30 dB at 1 kHz. The lower frequencies and the very high frequencies are devaluated compared to the middle frequency range when determining the dB(A)-value. As a matter of fact, the A-weighted level can also be determined from measured third-octave band levels. The levels given in Fig. 1.5 are added to the third-octave band levels and the total sound pressure level, now A-weighted, is calculated according to the law of level summation (1.4). The A-weighting function is standardized in EN 60651.
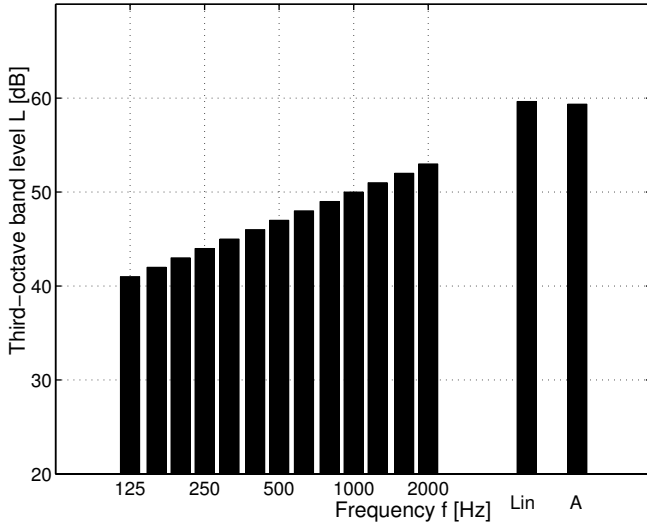
Fig. 1.6. Third-octave band, non-weighted and A-weighted levels of band-limited white noise

A practical example for the aforementioned level summation is given in Fig. 1.6 by means of a white noise signal. The third-octave band levels, the non-weighted (Lin) and the A-weighted (A) total sound pressure level are determined. The third-octave band levels increase by 1 dB for each band with increasing frequency. The linear (non-weighted) total sound pressure level is higher than each individual third-octave band level, the A-weighted level is only slightly smaller than the non-weighted level.

It should be noted that exceptions for certain noise problems (especially for vehicle and aircraft noise) exist, where other weighting functions (B, C and D) are used (see also Fig. 1.5). Regulations by law still commonly insist on the dB(A)-value.

Linearly determined single-number values, regardless of the filter used to produce them, are somewhat problematic, because considerable differences in individual perceptions do not become apparent. Fig. 1.4 clearly shows, for example, that 90 dB of level difference are needed at 1 kHz to increase the perception from 0 to 90 phon; at the lower frequency limit at 20 Hz only 50 dB

are needed. A simple frequency weighting is not enough to prevent certain possible inequities. On the other hand, simple and easy-to-use evaluation procedures are indispensable.

1.4 Noise fluctuating over time

It is easy to determine the noise level of constant, steady noise, such as from an engine with constant rpm, a vacuum cleaner, etc. Due to their uniform formations, such noise can be sufficiently described by the A-level (or third-octave level, if so desired.)

How, on the other hand, can one measure intermittent signals, such as speech, music or traffic noise? Of course, one can use the level-over-time notation, but this description falls short, because a notation of various noise events along a time continuum makes an otherwise simple quantitative comparison of a variety of noise scenarios, such as traffic on different highways, quite difficult. In order to obtain simple comparative values, one must take the mean value over a realistic average time period.

The most conventional and simplest method is the so-called 'energy-equivalent continuous sound level' L_{eq}. It reflects the sound pressure square over a long mean time

$$L_{eq} = 10 \lg \left(\frac{1}{T} \int_0^T \frac{p_{eff}^2(t)}{p_0^2} \right) dt = 10 \lg \left(\frac{1}{T} \int_0^T 10^{L(t)/10} \right) dt \qquad (1.5)$$

($p_0 = 20 \ 10^{-6} \ N/m^2$). Hereby $p_{eff}(t)$ indicates the time domain function of the rms-value and $L(t) = 10lg(p_{eff}(t)/p_0)^2$, the level gradient over time. The square of a time-dependent signal function is also referred to as 'signal energy', the energy-equivalent continuous sound level denotes the average signal energy; this explains the somewhat verbose terminology. For sound pressure signals obtained using an A-filter, third-octave filters, or the like, we use an A-weighted energy-equivalent level.

Depending on the need or application, any amount of integration time T can be used, ranging from a few seconds to several hours. There are comprehensive bodies of legislation which outline norms set by a maximum level L_{eq}, which is permissible over a certain time reference period ranging up to several hours. For instance, the reference time period 'night' is normally between the hours of 10 p.m. and 6 a.m., an eight-hour time period. Normally, for measuring purposes, a much smaller mean time frame is used in order to reduce the effect of background noise. The level L_{eq} is then reconstructed based on the number of total noise events and applied over a longer period of time. For example, let L_{eq} be an energy-equivalent sound level to be verified for a city railway next to a street. The mean time period for this measurement would then be approximately how long it takes for the train

to pass by the street. That is, we are looking for the energy-equivalent continuous sound level for an average period that corresponds to the time it takes for the train to pass by the street, such as of 30 seconds $L_{eq}(30s)$. Suppose the train passes by every 5 minutes without a break. In this case, we can easily calculate the long-term L_{eq} (measured over several hours, as may correspond to the reference periods 'day' or 'night', for instance) by $L_{eq}(long) = L_{eq}(30s) - 10lg(5min/30s) = L_{eq}(30s) - 10\,dB$.

Applying mean values is often the most sensible and essential method for determining or verifying maximum permissible noise levels. However, mean values, per definition, omit single events over a time-based continuum and can thus blur the distinctions between possibly very different situations. A light rail train passing by once every hour could well emit similar sound levels L_{eq} as characterized by the permanent noise level of a busy street over a very long reference period, for example. The long-term effects of both sources combined may, in fact, result in one of the sources being completely obliterated from the L_{eq} altogether (refer to Exercise 5).

The energy-equivalent continuous sound level serves as the simplest way to characterize sound levels fluctuating over time. Cumulative frequency levels can be determined by the peak time-related characteristic measurements, a method which is instrumental in the statistical analysis of sound levels.

1.5 Summary

Sound perception is governed by relativity. Changes are perceived to be the same when the stimulus increases by a certain percentage. This has led to the conclusion of the Weber-Fechner law, according to which perception is proportional to the logarithm of the stimulus. The physical sound pressures are therefore expressed through their logarithmic counterparts using sound levels of a pseudo-unit, the decibel (dB). The entire span of sound pressure relevant for human hearing, encompassing about 7 powers of ten, is reflected in a clearly defined scale from about $0\,dB$ (hearing threshold) up to approximately $140\,dB$ (threshold of pain). A-weighting is scaled to the human ear in order to roughly capture the frequency response function of hearing. A-weighted sound levels are expressed in the pseudo-unit $dB(A)$.

Sounds at intermittent time intervals are quantified using mean time values. One such quantification method approximates 'energy-equivalent permanent sound levels.'

1.6 Further reading

Stanley A. Gelfand's "Hearing – an Introduction to Psychological and Physiological Acoustics" (Marcel Dekker, New York 1998) is a physiologically oriented work and contains a detailed description of the anatomy of the human ear and the conduction of stimuli.

1.7 Practice exercises

Problem 1

An A-weighted sound level of $50\,dB(A)$ originating from a neighboring factory was registered at an emission control center. A pump is planned for installation $50\,m$ away from the emission control center. How high can the A-weighted decibel level, resulting from the pump alone, be registered at the emission control center so that the overall sound level does not exceed $55\,dB(A)$?

Problem 2

A noise contains only the frequency components listed in the table below:

f/Hz	$L_{Thirdoctave}/dB$	Δ_i/dB
400	78	-4,8
500	76	-3,2
630	74	-1,9
800	75	-0,8
1000	74	0
1250	73	0,6

Calculate

- both non-weighted octave levels
- the non-weighted overall decibel level and
- the A-weighted overall decibel level.

The corresponding A-weights are given in the last column of the table.

Problem 3

A noise consisting of what is known as white noise is defined by a $1\,dB$-increase from third-octave to third-octave (see Figure 1.6). How much does the octave level increase from octave to octave? How much higher is the overall decibel level in proportion to the smallest third-octave when N third-octaves are contained in the noise? State the numerical value for $N = 10$.

Problem 4

A noise consisting of what is known as pink noise is defined by equal decibel levels for all the third-octaves it contains. How much does the octave level increase from octave to octave and what are their values? How much higher is the overall decibel level when there are N thirds contained in the noise? State the numerical value for $N = 10$.

Problem 5

The energy-equivalent permanent sound level is registered at $55\,dB(A)$ at an emission control center near a street during the reference time period 'day,' lasting 16 hours. A new high-speed train track is scheduled to be constructed near the sight. The 2-minute measurement of a sound level L_{eq} of a passing train is $75\,dB(A)$. The train passes by every 2 hours.

How high is the energy-equivalent permanent sound level measured over a longer period of time (in this case, reference time interval 'day')

- a) of the train alone and
- b) of both sound sources combined?

Problem 6

A city train travels every 5 minutes from 6 a.m. to 10 p.m. At night, between 10 p.m. and 2 a.m., it travels every 20 minutes, with a break from 2-6 a.m. in between. A single train passes within 30 seconds and for this time duration, the sound level registers at $L_{eq}(30s) = 78\,dB(A)$. How high is the energy-equivalent permanent sound level for the reference time intervals 'day' and 'night'?

Problem 7

The sound pressure level L of a particular event, such as the emission of a city train as in the previous example, can under certain circumstances only be measured against a given background noise, such as traffic. Assume that the background noise differs from the sound event to be measured at a decibel level ΔL. How high is the actual combined noise level? State the general equation for measuring errors and the numerical value for $\Delta L = 6\,dB$, $\Delta L = 10\,dB$ and $\Delta L = 20\,dB$.

Problem 8

As in Problem 7, sound emission is to be measured in the presence of a noise disturbance. How far away does the the noise source have to be in order to obtain a measuring error of $0.1\,dB$?

Problem 9

Sometimes filters of a relatively constant bandwidth are employed to take finer measurements in sixth-octaves increments as opposed to the customary third-octave increments. State the equations for

- the consecutive center frequencies
- the bandwidth and
- the limiting frequencies.

Problem 10

In a calculated measurement where an octave and all of its constituent thirds are given, it appears that one of the thirds may have been a measurement error. How can the result be checked against the other three values that are assumed to be correct?