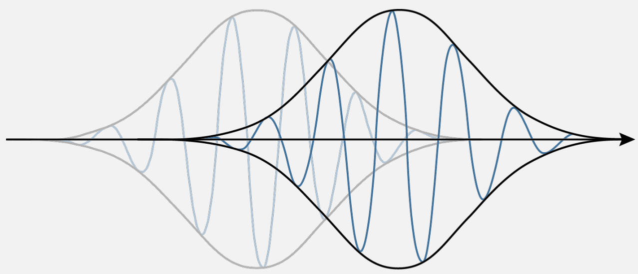
HEAD acoustics

Acoustics Glossary



Glossary of terms used in the fields of acoustics

Information on this document

Contents

This document includes a glossary of terms used in the fields of sound and vibration acquisition, analysis and evaluation. The terms are sorted alphabetically with cross-references in the respective explanations.

Target group

This glossary is intended both for those who need a reference work with short and concise definitions of everyday acoustic terms and for those who wish to familiarize themselves with the subject of sound and vibrations.

Questions?

Do you have questions? You miss an important term? Your feedback is appreciated!
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Acoustics Glossary

Absorption: see Sound absorption

Accelerometer: Accelerometers are used to measure the accelerations (vibrations) of measurement objects. In most cases piezoelectric accelerometers are used. Using these sensors, an electric charge is generated by the acceleration of a mass coupled to a piezoelectric element. An integrated circuit converts the charge into a voltage signal that can be detected by a recording device. Piezoelectric accelerometers are particularly suitable for measuring signals where the acceleration over time is changing relatively quickly.
For a correct measurement of acceleration, the coupling of the accelerometer to the measuring object is essential. In practice, various methods are available, e.g., magnet, wax, adhesive and screwing. The choice of coupling has a direct effect on the frequency range that can be measured with the sensor.

Figure 1: Accelerometer

ADAT interface: ADAT (Alesis® Digital Audio Tape) was developed by Alesis Corp. The ADAT interface is used for the digital multi-channel transmission of audio signals. Up to eight data channels are optically transmitted via optical fibers with TOSLINK connectors.

A/D converter or analog to digital converter, ADC: An A/D converter is a component used to convert an analog input signal into a digital signal. This kind of conversion, or digitization, is necessary to store the signal digitally and to analyze and modify it by using computer-aided signal processing. The sampling rate, the resolution and the quantization are important parameters for digitization.

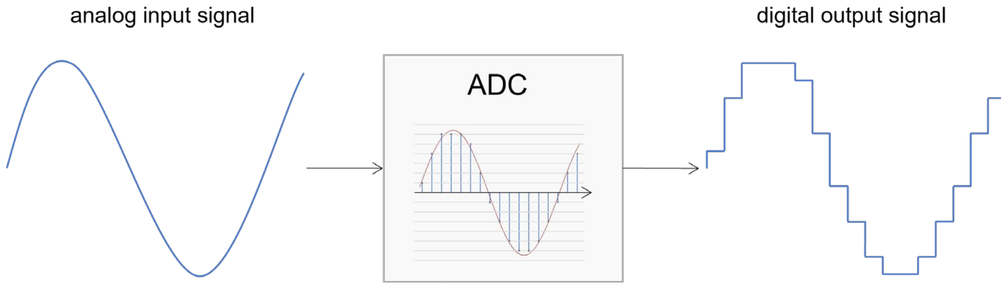


Figure 2: Analog to digital conversion

AES/EBU (Audio Engineering Society/European Broadcasting Union): AES/EBU refers to the specification of an interface for the transmission of digital one-channel or two-channel audio signals. Various types of cables can be used for transmission, e.g., balanced cables with an impedance of *110 ohms* and XLR connectors or unbalanced coaxial cable with an impedance of *75 ohms* and BNC connectors.

Airborne noise: Airborne noise is the noise that propagates through the air. A vibrating sound source surrounded by air excites neighboring air molecules, causing them to vibrate and, in turn, excite neighboring air molecules. The vibrations thus continue as longitudinal waves in the air. The human ear is able to perceive airborne noise with a frequency of about 20 Hz to 20 kHz.

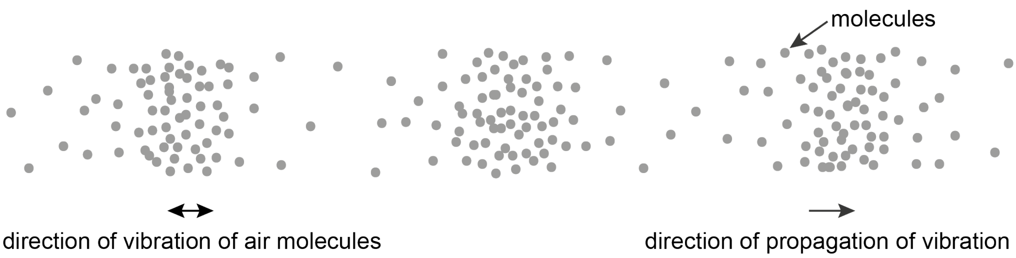


Figure 3: Propagation of airborne noise

Aliasing: Aliasing effects can occur when analog signals are digitized. If the signal to be sampled includes frequency components that are higher than half the sampling rate f_s ($f > \frac{f_s}{2}$), these are interpreted as lower frequencies during the digitalization (see also Sampling theorem). However, these spurious low-frequency components were not included in the original signal and thus are disturbing noises that overlap the useful signal. See figure: The high-frequency (red) time signal was sampled (black dots) at too low a sampling rate, resulting in the low-frequency (blue) disturbing noise

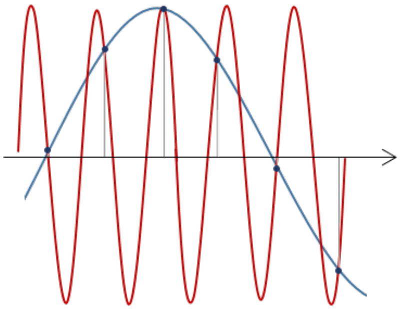


Figure 4: Aliasing

In order to avoid such aliasing effects, the input signal should be filtered by a low-pass filter (anti-aliasing filter) before digitalization.

Analog signal: An analog signal is a continuous and uninterruptible signal that describes the continuous time response of a physical quantity such as sound pressure. In contrast to a digital signal, an analog signal can theoretically take on an infinite number of values and thus has a continuous history

Anechoic chamber: An anechoic chamber is a room whose wall covering is designed in such a way that disturbing reflections only occur at wavelengths equal or longer than a certain wavelength. Sound reflection is reduced by covering the walls with sound absorbing materials. These materials convert the sound energy of the incident sound into thermal energy. Since the walls, ceiling and floor do not reflect the sound, the sound field in the anechoic chamber equals a free field. For this reason, an anechoic chamber is also called a free-field room.

Artifical head (also dummy head): An artificial head is used for binaural sound recording. It consists of a head replica, in each ear of which a microphone is placed (usually at or a short distance inside the entrance of the ear canal). The current artificial head models from HEAD acoustics have a simplified outer geometry. This geometry affects the sound field in the same way as a person with average external geometry does.

Figure 5: Artificial head HMS IV

Auditory event (also hearing event): If a physical sound event is perceived auditorily by a human being, it is called an auditory event. An auditory event can be characterized by different properties (e.g., the timbre or the temporal structure). The perception of various sound events as auditory events is investigated in psychoacoustics, for example, by means of listening tests.

Auditory sensation area: The auditory sensation area is a range of frequency and sound pressure level that can be perceived by the human ear. Downward in level, the auditory sensation area is limited by the hearing threshold and upwards by the threshold of pain.

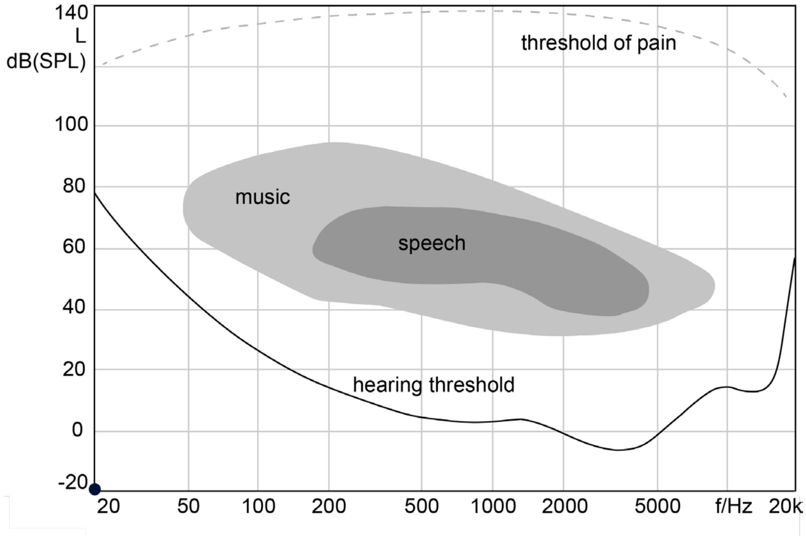


Figure 6: Auditory sensation area

Hearing threshold: The hearing threshold is the lowest sound pressure level that can just be perceived. The hearing threshold is frequency-dependent and rises sharply at high and low frequencies. The highest sensitivity of the human ear is between 2 *kHz* and 5 *kHz*. Signals at lower or higher frequencies must have a significantly higher sound pressure level to be still perceptible. The hearing threshold can change over the course of a lifetime. It is especially the high frequencies that are perceived less and less by elderly people. In addition, the sound field affects the course of the hearing threshold. ISO 389 defines different hearing thresholds for free field and diffuse fields.

Threshold of pain: The threshold of pain is the sound pressure level that is perceived as being painful by the listener. Due to the high level of stress for the test person and possible hearing damage, the threshold of pain cannot be determined unambiguously. It is less dependent on frequency than the hearing threshold and ranges approximately between *120 dB* and *140 dB,* thus corresponding to sound pressures between *20* and *200 Pa*.

Aurally-accurate: The transmission of sound signals is aurally-accurate if it is accurate in terms of perceived reproduction of level, direction, distance and timbre. For this purpose, the sound pressure signals arising at the eardrum when listening to a recording must be comparable to the sound pressure signals that would have arisen at the eardrum in the original sound field.

Aurally-adequate or hearing-related: Aurally-adequate analyses include a signal processing that is modelled on human hearing. For this purpose, algorithms are developed and used that model the relationship between acoustic stimuli and the sensory impressions induced in humans. By means of aurally-adequate analyses, the human auditory sensation, which is the decisive factor in the evaluation of sound events, can thus be taken into account.

Auricle: The auricle is the part of the human ear that lies on the outside of the head. The outer edge of the auricle is called the helix, the inner edge, which is closer to the ear canal, is called the anti-helix. The depression of the auricle in front of the ear canal entrance is called cavum conchae.

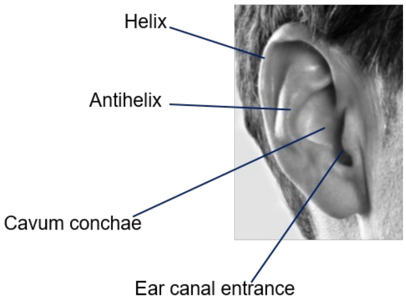


Figure 7: Various parts of the auricle

AVAS (Acoustic Vehicle Alerting System): AVAS is an exterior acoustic warning system that is installed in low-noise electric vehicles. The artificially generated sounds are emitted at low speeds to warn road users. At higher speeds, the tire noise and wind noise are loud enough, that no artificial warning sounds are required. In many countries, acoustic warning systems for electric and hybrid vehicles are required by law, although the specifications vary. For example, the speed limit up to which the AVAS must be active is 20 km/h in Europe and China, and 30 km/h in the US.

Averaging: Averaging is used to determine an average value of a defined set of numbers. There are different ways of calculating the average value. The best known method for calculating an average is **arithmetic averaging**, for which all values of the set of numbers are added together and divided by the number of values: $\frac{1}{n}\sum_{i=1}^{n} x_i$.

When averaging sound pressure levels arithmetic averaging must not be applied directly, since the sound pressure level is a logarithmized ratio. The sound pressure levels must first be converted into sound pressures which are then arithmetically averaged and reconverted to a sound pressure level. This type of averaging is also referred to as energetic averaging.

Further averaging methods are:

Root mean square: For the root mean square averaging, the squares of all values are summed up and divided by their number. Then the square root is calculated for this value: $\sqrt[2]{\frac{1}{n}\sum_{i=1}^{n} x_i^2}$.

The root mean square value evaluates the larger numbers of the set more strongly than the arithmetic mean value.

Geometric averaging: In geometric averaging, a mean value is determined using the n^th root of the product of all values in the set of numbers: $\sqrt[n]{\prod_{i=1}^{n} x_i}$. The geometric mean is less than or equal to the arithmetic mean. This type of averaging gives less weight to the larger numbers in a set of numbers.

Moving average: The moving average is calculated iteratively for a defined section. Then the considered section is moved in an overlapping way, i.e., the first value is deleted from the considered section, and instead the first value after the section is included and a new average value is calculated, etc. Moving averaging is used, for example, to smooth time signals. In many cases, an additional weighting is applied when calculating the moving average, thus achieving a weighted smoothing of the data series.

Exponential moving average: Exponential averaging weights the data points of a time signal exponentially decreasing. In this way, more recent data points are more important than those dating further back. Due to this kind of weighting, strong deflections of individual data points are increasingly attenuated the earlier they date.

A weighting: see Sound pressure level

Bark: see Critical bands

Beamforming: Using beamforming algorithms, such as the delay and sum algorithm, microphone arrays can be used for sound source localization. Due to the different path length from a sound source to the individual microphones of the array, there are phase and amplitude differences in the recorded time signals. These phase and amplitude differences can be calculated for points at a defined distance in front of the array. The calculated differences are used to correct the measured microphone signals for each point. In a subsequent summation, the proportions of sound sources actually located at this point add up constructively, while proportions from sources at other locations are almost cancelled out.

The corrected and summed time signals can, for example, be subjected to a level analysis. The level can then be superimposed on an image of the sound source in a color-coded form. Viewing these source maps allows intuitive sound source localization and effective sound optimization.

Figure 8: Sound source localization by means of beamforming

Beat: A beat occurs when two sine oscillations whose frequency differ only slightly are superimposed. Due to the small frequency difference, the phase position of the two oscillations is not constant. This leads to the fact that the instantaneous values of the two oscillations sometimes rise and sometimes fall and so the resulting oscillation has a rising and falling amplitude. The human ear perceives only one tone, the volume of which increases and decreases periodically.

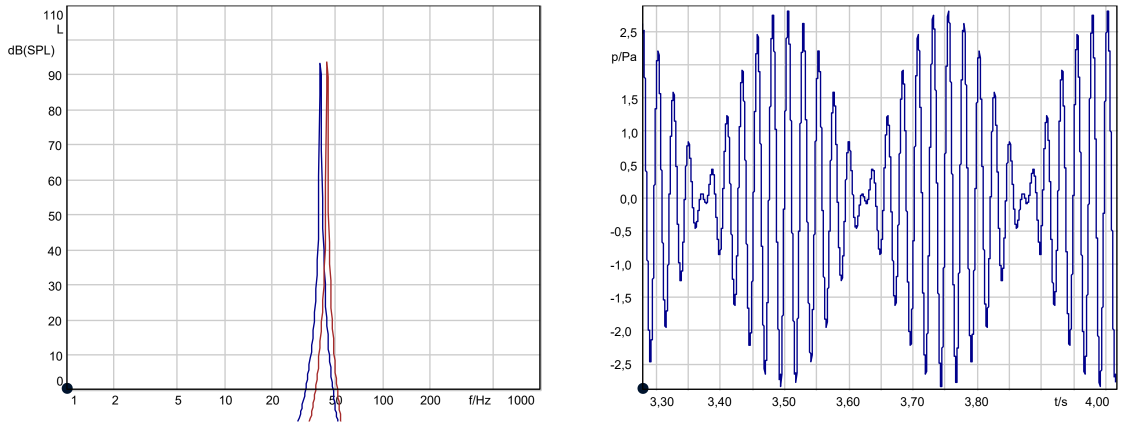


Figure 9: Frequency spectrum of two sinusoidal tones with small frequency difference (left); time signal of the resulting oscillation with beat (right)

Binaural: with both ears, concerning both ears
The expression “binaural” is formed by two Latin words: "bi" for two and "auris" for ear.

Binaural hearing: During binaural hearing, the hearing organ evaluates the interaural differences, for example. Binaural hearing allows

- identify the origin of a sound source,
- separate sound sources from different directions and
- partially suppress unwanted noises.

Binaural measurement technology: Binaural measurement technology allows the listener to perceive the sound situation, when listening to a recording, as if he were in the original sound field. This is possible because binaural recordings contain the signal components that are necessary for binaural hearing. Binaural recordings can be recorded with an artificial head, for example.

Figure 10: Artificial head HMS IV

BNC connection: The BNC connector is used to connect coaxial cables. BNC plugs are equipped with a bayonet lock which prevents accidental disconnection.

Figure 11: BNC connector

CAN (Controller Area Network): The CAN bus is a serial bus system that networks the various control units of a vehicle. It was developed to reduce wiring harnesses and thus save costs and weight. The CAN bus consists of a two-wire data line to which all components are connected via short stubs. The standardization of the communication between the control units by means of the CAN protocol also enables function monitoring. With the CAN bus, data is transmitted by the bus levels CAN_High and CAN_Low. The signal of the CAN_Low line is redundantly inverted to that of the CAN_High line. This opposite potential change leads to a high electrical noise immunity.

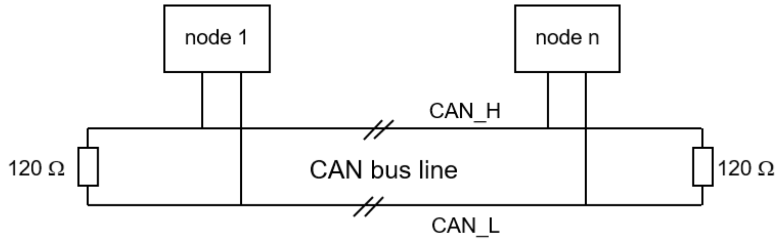


Figure 12: Schematic illustration of a CAN bus

The CAN bus data is transmitted in a manufacturer and vehicle-specific form. Therefore, a corresponding database is required for extracting a specific piece of information from the data stream. By default, these are managed with database software from Vector Informatik (USA: Vector CANtech Inc., Japan: Vector Japan Co. Ltd.) and stored in a file in DBC format.

CAN FD (CAN with Flexible Data Rate): CAN FD is an extension of the original CAN bus protocol with flexible data rate and was developed to meet the increased demands on data accuracy, transfer rates and cycle times. CAN FD significantly increases the usable data rate and the payload length.

Cavum conchae: see Auricle

Center frequency: The center frequency f_m is the geometric mean of the lower (f_u) and upper cut-off frequency (f_o) of a frequency band: $f_m = \sqrt[2]{(f_u \cdot f_o)}$
On a logarithmic frequency scale, the center frequency is halfway between the two cutoff frequencies. With a band pass filter, the transfer function reaches its maximum at the center frequency.

Cinch connection (also RCA connection): The cinch connection is used to connect coaxial cables. The transmission is unbalanced and therefore very susceptible to interference.

Figure 13: Cinch connector

Coaxial cable (or coax cable): Coaxial cables are two-pole cables with an inner and an outer conductor. The structure of the cable is concentric, thus the outer conductor surrounds the inner conductor. There is an insulating layer between the inner and the outer conductors. Due to the concentric structure, the outer conductor shields the signal-carrying inner conductor against external interference radiation. Cinch or BNC connectors are often used for coaxial cables.

Cocktail party effect: The cocktail party effect is a feature of the human auditory system that enables us to concentrate on a single sound source in noisy environments such as a cocktail party. With the help of binaural signal processing, a single source is selected from a number of signal sources and the disturbing background noise is suppressed. In this way, speech intelligibility, for example, can be significantly improved over that achieved with one-eared hearing, or auditioning a recording made with a single microphone.

Coherence: Coherent sound waves have a constant phase relationship to each other. The coherence of sound waves is the prerequisite for the persistent occurrence of interference. Sound sources the phase relationship of which is not constant are referred to as incoherent.

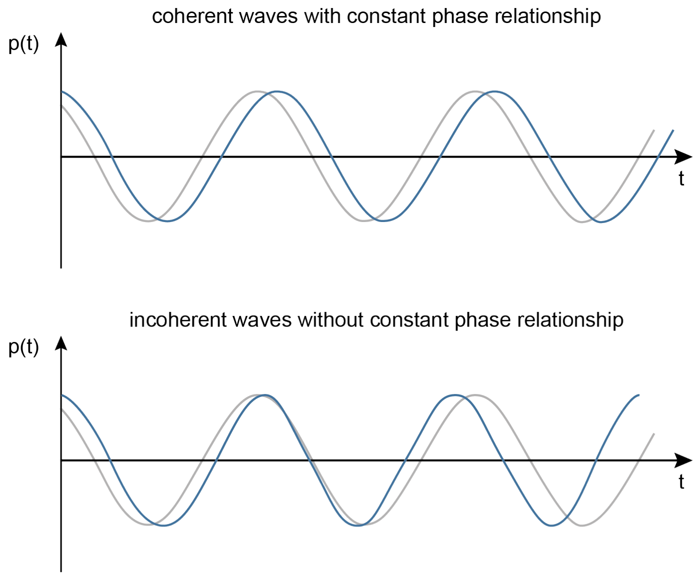


Figure 14: Coherent and incoherent sound waves

In signal analysis, the coherence function shows the linear dependence of two signals. The coherence function can be used to determine, for example, the similarity between input and output signals after they have passed through a transmission path. A high degree of coherence means a high dependence between the input and the output signal. Disturbing signals such as noise and non-linear distortions reduce the coherence between the input and the output signal, thus resulting in a low degree of coherence.

Critical bands: Various experiments and listening tests have shown that the human hearing combines sound stimuli whose frequencies are close together, into frequency bands. These frequency bands are called critical bands. The audible frequency range was divided into 24 critical bands with the unit *Bark* by K. E. Zwicker. The width in Hz of the critical bands increases with the frequency.

Critical band [Bark]	Frequency [Hz]
0	0
1	100
2	200
3	300
4	400
5	510
6	630
7	770
8	920
9	1080
10	1270
11	1480
12	1720
13	2000
14	2320
15	2700
16	3150
17	3700
18	4400
19	5300
20	6400
21	7700
22	9500
23	12000
24	15500

Table 1: Critical bands according to Zwicker

The ERB (Equivalent Rectangular Bandwidth) scale describes another division of the critical bands. The width of the critical bands on the ERB scale is narrower than that of the critical bands on the Bark scale.

Crosstalk (or sometimes x-talk): The term crosstalk stands for the undesired influence of actually independent signal channels.
With electrical crosstalk, the signal of a cable can, for example, be coupled inductively to the signal of another cable. Electrical crosstalk can be reduced by appropriate shielding of the cables.
Another example where unwanted crosstalk occurs is the playback of binaural recordings using two loudspeakers. In contrast to playback with headphones, the radiated sound fields overlap during loudspeaker playback. As a result, the right ear also receives signal components that were originally intended for the left ear and vice versa. Digital filters can be used to compensate for crosstalk for a defined position in the playback room.

D/A converter: A D/A (digital to analog) converter converts discrete information of the digital signal into an analog signal that can be continuously provided to an analog operating device (e.g., a loudspeaker). D/A converters are integral components of digital entertainment and communication electronics.

Daisy chain: Daisy-chain cabling is used to connect the hardware components in series (one behind the other).

Data transmission rate or bit rate: The data transmission rate indicates the amount of data that can be digitally transmitted by a system within a time unit. The data transmission rate is measured in *bits per second* (*bit/sec*) or in *kilobits, megabits, gigabits per second*, etc. for larger amounts of data.

Decibel: *Bel* is an auxiliary unit. It denotes the decadic logarithm of the ratio of two similar quantities. In the field of acoustics, the quantities used often range in values of several decimal powers (e.g., the sound pressure that can be perceived by humans from about 10^{-5} *Pa* to 10^{2} *Pa*). The specification as a logarithmic ratio reduces the value range and simplifies the presentation and interpretation of the quantities (see also sound pressure level). The specification as logarithmic quantity is not only used for the sound pressure, but also for the sound intensity and the sound power. As a rule, *decibel (dB)* is used, i.e., the tenth part of a Bel.

Degree of freedom (DOF): In general, the degrees of freedom represent the number of parameters of a system that can be varied independently. In the field of mechanical engineering, degree of freedom describes the number of independent movement possibilities of a system. A free point generally has six degrees of freedom: three degrees of freedom of translation (x, y, z direction) and three degrees of freedom of rotation (roll, pitch, yaw).

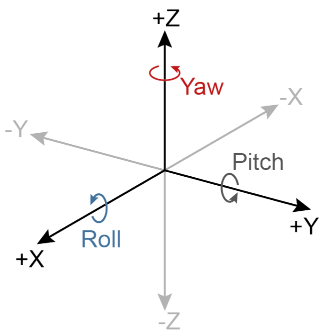


Figure 15: Six degrees of freedom

Dichotic: In dichotic sound presentations (e.g., via headphones) the signals for the left and right ear are different (in contrast to diotic sound presentations). With binaural recordings, the signals are dichotic.

Diffraction of sound waves: When plane waves impinge upon an obstacle that is equal to or smaller than their wavelength λ, spherical waves are created. These spherical waves propagate evenly in all directions. Thus, the sound is diffracted by the obstacle, allowing the sound to reach the geometric sound shadow of the obstacle and be heard.

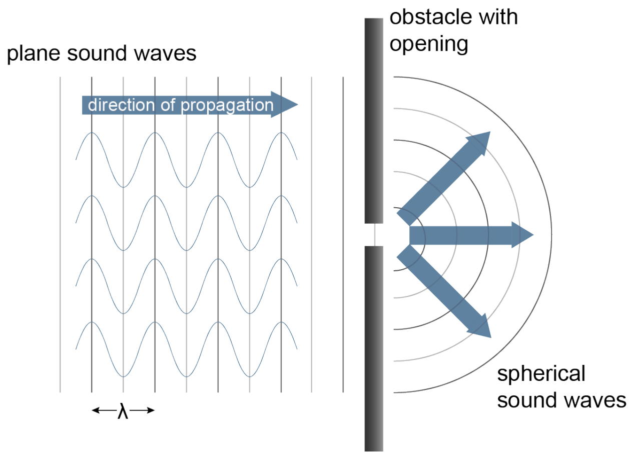


Figure 16: Diffraction of sound waves at a circular hole

Diffuse field: see Sound field

Diffuse field equalization: The diffuse field equalization is a digital filter that can be used to modify a binaural recording to be compatible with a measurement microphone recording. A high degree of compatibility is only achieved, however, if there really was a diffuse sound field present during the recording, i.e., a sound field with equally probable sound incidence from all directions.

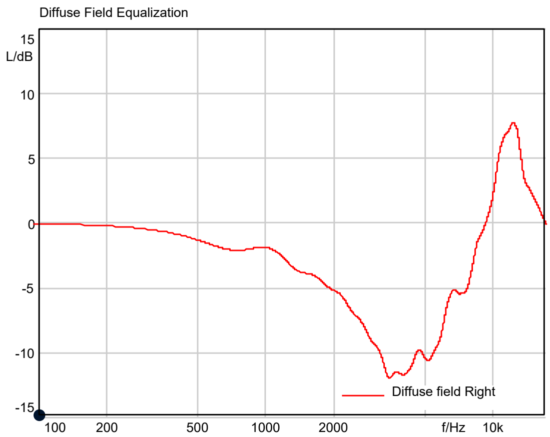


Figure 17: Diffuse field equalization for an artificial head measurement system HMS IV (right ear)

Diffusion: Diffusion is a room acoustic characteristic. It is a measure of the intensity distribution of the reflected sound of a source in a room. Rooms with a balanced diffusion often provide a better listening experience. A higher variety of reflection possibilities, e.g., curved or angled surfaces, increases a room's diffusion.

Digital signal: A digital signal is a signal that is discrete in both value and time. A digital signal is generated from an analog signal by sampling it at given times (thus discrete in time). The frequency of this sampling is determined by the sampling rate. In addition, the digital signal can only take on graded values (thus discrete in value). The resolution defines how many different values the signal can take on. An analog signal is digitized using an A/D converter.

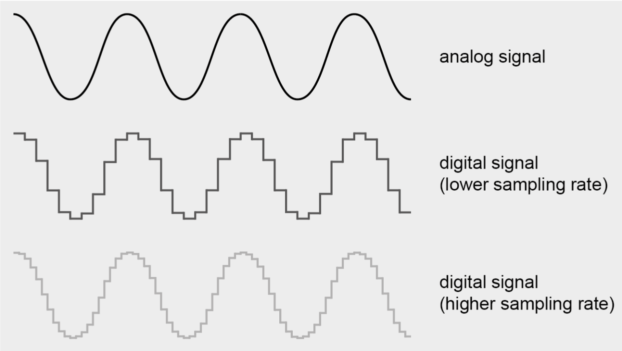


Figure 18: Digital signal – sampled with lower or higher sampling rate

Digital signal processor, DSP: A digital signal processor processes digital signals by means of digital signal processing. DSPs are used in digital entertainment and communication electronics, e.g., in sound cards and MP3- players.

Diotic: In diotic sound presentations (e.g., via headphones) the signals for the left and right ear are identical (in contrast to dichotic sound presentations).

Direct sound: Direct sound is the sound that directly (i.e., without reflections) reaches the receiver (e.g., the listener's ear) first. Direct sound has a significant influence on the directional determination of human hearing.

Distortion: Distortion changes an input signal so that the output signal no longer fully corresponds to the input signal. A distinction is made between linear and non-linear distortions:
linear distortion: With linear distortion, the original signal shape is not changed, but only its amplitude. A frequency-dependent attenuation of a signal is an example of linear distortion.
non-linear distortion: With non-linear distortion, the signal shape is changed, thus creating frequencies that were not present in the original signal. Non-linear distortion occurs, for example, when a recording is overloaded. The distortion factor is a measure of nonlinear distortions.

Distortion factor: The distortion factor is a measure of non-linear distortions of an originally pure sine wave. It includes unwanted noise components whose frequencies are integer multiples of the fundamental frequency and describes their share in the overall signal. The distortion factor is given as a ratio in percent. It is often equated with the THD (Total Harmonic Distortion). However, this is not quite correct:
Distortion factor: ratio of harmonics to overall signal
THD: ratio of harmonics to fundamental frequency
The distortion factor and the THD are therefore not identical, but are comparable for small values.

Doppler effect: The Doppler effect during sound transmission occurs when the sound source, the receiver or both move in relation to the transmission medium. This causes the distance between the sender and the receiver of a signal to change and the signal to be compressed or stretched. This compression or stretching changes the wavelength and thus the frequency. If the distance between sender and receiver decreases, the frequency for the receiver increases (see figure). If the distance between sender and receiver increases, the frequency for the receiver decreases.

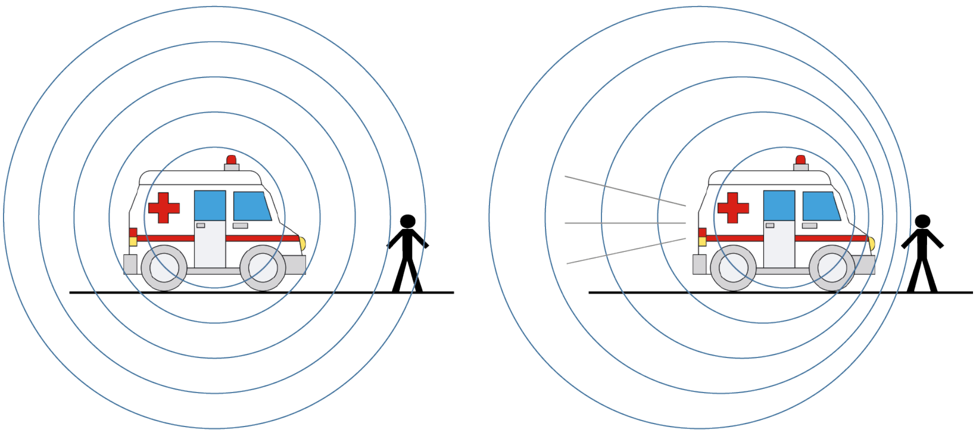


Figure 19: Sound propagation with a stationary vehicle (left), sound propagation with a moving vehicle (right)

D-Sub connection: The D-Sub connection is a widely used connector system for serial interfaces. To ensure a secure connection, the D-Sub connector can often be screwed to the socket. The 9-pin D-Sub connector is used, for example, as an interface for recording information from a CAN bus.

Figure 20: 9-pin D-Sub connection

Dynamic range: The dynamic range indicates the usable amplitude range in which the useful signal can range. At the bottom, the dynamic range is limited by the noise floor. Signals with a lower sound pressure level are masked by the noise floor of the system. Upwards, the dynamic range is limited by reaching the maximum passable or encodable level. An overload situation occurs if the sound pressure level of a signal exceeds this limit. There is no standardized calculation method for determining the dynamic range of a measurement system, for example. The dynamic range determined is highly dependent on the analysis bandwidth used. When specifying the dynamic range, the analysis bandwidth should therefore always be included. The term dynamic range is sometimes used erroneously as a synonym for the tone-to-noise ratio.

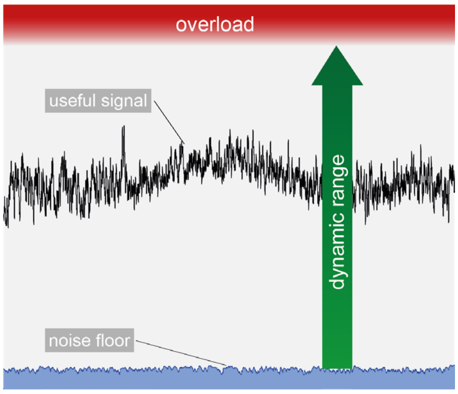


Figure 21: Dynamic range (schematic representation)

Echo: An echo is a discrete sound reflection that reaches the listener with significant delay. Due to the delay, the listener no longer hears the reflection as a reverberation of the original sound. Instead, the reflection is perceived as a separate auditory event. An echo only occurs when the reflected sound has been sufficiently delayed. The required time interval between the and the reflection is called echo threshold. Echo thresholds do not have fixed values, but depend on both the tonal character of the sound and the level of the reflection. If the sound is reflected with a shorter delay than the echo threshold, it is perceived as fused with the original sound or as part of reverberation.

Envelope: The envelope is the curve which connects the maxima of a periodic oscillation. If the signal is modulated in sinusoidal form, the envelope is a sine wave. With an unmodulated sine wave signal, the envelope is a straight, level line.

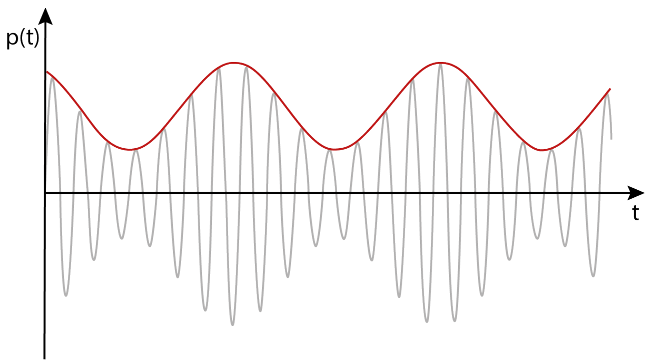


Figure 22: Amplitude-modulated sinusoidal tone (grey) with envelope (red)

Equalization: Equalization is a filter, typically implemented digitally, that changes the timbre of a sound. The equalization of a measurement signal serves to linearize the frequency response (see also: Free field equalization, Diffuse field equalization).

Equivalent continuous sound level (L_{eq}): The equivalent continuous sound level is a measure of the average noise exposure resulting from a long-lasting, time-varying noise situation. In contrast to the sound pressure level indicating the level of a sound at a certain instant in time, the equivalent continuous sound level is a time-averaged quantity. The equivalent continuous sound level corresponds to the sound pressure level of a noise with constant amplitude, which transmits the same sound energy as the sound with the time-varying amplitude for the same exposure time. The equivalent continuous sound level is usually given in the unit *dB(A)* and is always linked to a defined period of time.

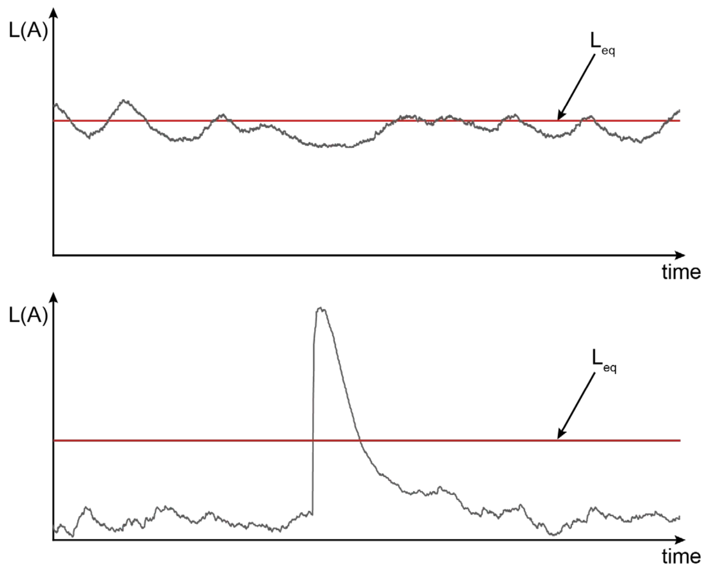


Figure 23: Equivalent continuous sound levels of a constant sound (top) and a transient sound (bottom)

Excitation: Excitation is the supply of energy that sets an oscillatory system into vibration.

Fade: Using a fade reduces noise such as clicks at the beginning and end of a signal that may occur when playing back sound files.
Fade-in: The sound pressure level of the signal is increased over a certain period of time.
Fade-out: The sound pressure level of the signal is reduced over a certain period of time.
Crossfade: Crossfade is an overlapped combination of fade in and fade out and can be used when a sound file is repeated multiple times or when different files are played back directly one after another.

Filter: A filter changes the amplitude or phase of a signal depending on its frequency. The transfer function of a filter describes its effect on the input signal. In analog circuits, filters can be implemented using electrical resistors, coils and capacitors. If the signal is available in digital form, digital filters are used. Digital filters can be realized by means of a sequential program with a signal processor. They are divided into IIR filters and FIR filters.
IIR filter: IIR filters are filters with infinite impulse response. These filters always contain a feedback component (recursive filters). An advantage of the IIR filters are higher Q factors (i.e., stronger selectivity) that can be achieved. In addition, IIR filters have a relatively low absolute group delay (latency). However, no linear phase response is possible for IIR filters, i.e., different frequencies pass through the filter at different speeds.
FIR filter: FIR filters are filters with a finite impulse response. In most cases they are filters without feedback component (non-recursive filters). FIR filters can be implemented in linear phase, i.e., different frequencies pass through the filter at the same speed. The absolute group delay (latency) is relatively large.

Furthermore, filters can be divided according to their transmission characteristics:
High-pass filter: High-pass filters only allow the high frequencies contained in a signal to pass. Frequencies below a certain cut-off frequency f are attenuated or filtered out of the signal by a high-pass filter. High-pass filters can be used, for example, to reduce DC voltage components.
Low-pass filter: Low-pass filters only allow the low frequencies contained in a signal to pass. Frequencies above a certain cut-off frequency f are attenuated or filtered out of the signal by a low-pass filter. Low-pass filters are used, for example, to avoid aliasing.

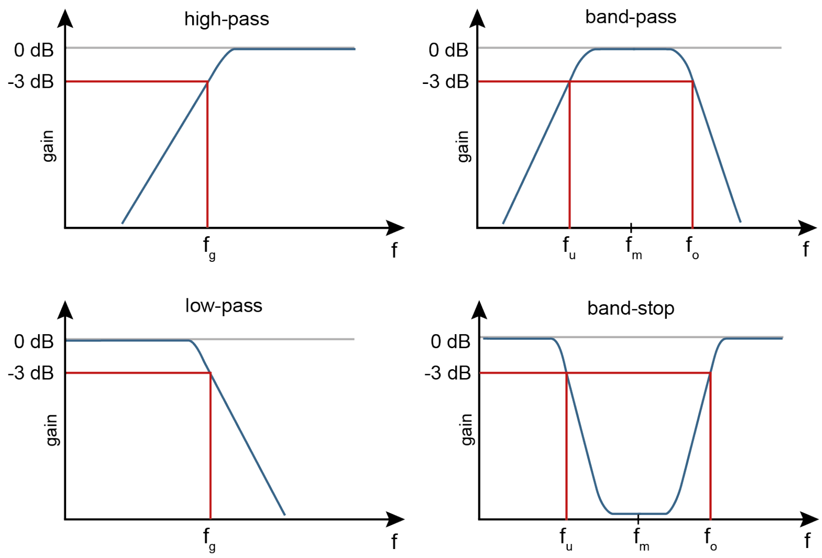


Figure 24: Transmission characteristics of high-pass, low-pass, band-pass and band-stop filters (schematic)

Band-pass filter: Band-pass filters only allow frequencies within a certain frequency range (frequency band around the center frequency f_c) to pass. Frequencies below and above the passband are attenuated or filtered out of the signal.
Band-stop filter: Band-stop filters allow all frequencies to pass except those of a certain frequency range (frequency band around the center frequency f_c). Frequencies within the passband are attenuated or filtered out of the signal. Band-stop filters can be used to attenuate or remove disturbing noise with a constant frequency from a signal.

The transfer function of a filter is determined by the passband and the stopband. All frequencies within the passband are to be transmitted as undamped as possible, while the frequencies of the stopband are to be suppressed. The cut-off frequency forms the border between the passband and the stopband. With a band-pass or band-stop filter, the difference between the upper and lower cut-off frequency (f_u, f_l) represents the bandwidth of the filter.

$$Band\ width = upper\ cut\text{-}off\ frequency\ f_u - lower\ cut\text{-}off\ frequency\ f_l$$

Filters are quantified using the filter quality and filter order:
Q factor: The filter quality Q describes the selectivity of a filter. The Q factor of a band-pass or band-stop filter is indicated by the quotient of the center frequency and the bandwidth of the filter:

$$Q = \frac{center\ frequency}{band\ width}$$

With a constant center frequency, a high Q factor thus means a small bandwidth (the filter works very selectively), while a low Q factor means a large bandwidth (the filter influences a large frequency range).
Filter order: The filter order is related to the slope of the transfer function of a filter. The slope marks the transition between the passband and the stopband. The higher the slope, the faster the transition occurs. For first-order high-pass or low-pass filters, the slope is *6 dB/octave*. With a second-order filter the slope is *12 dB/octave*, and so on. Note that the steeper the filter order, the steeper the filter slopes. In addition, the higher the filter order, the more the passband fluctuates. These ripples are an undesirable side effect of real filters at high filter orders.

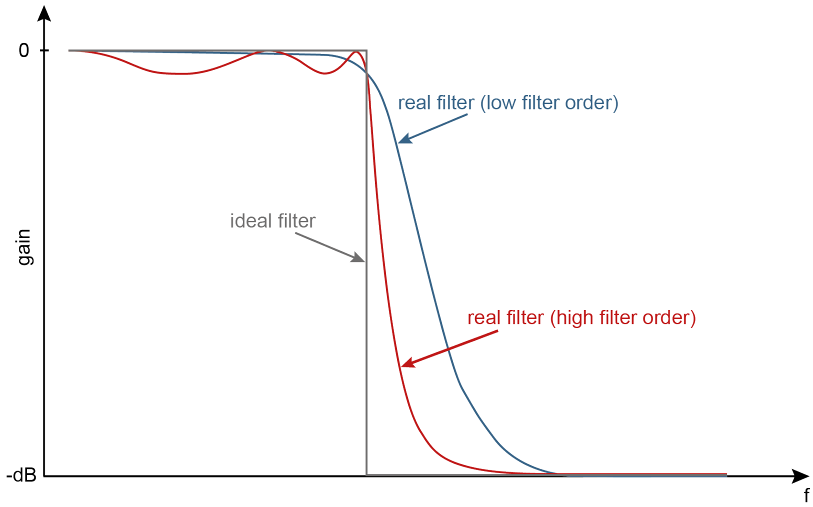


Figure 25: Filter with different filter orders (schematic)

Fluctuation strength: The sensation of fluctuation strength results from signal fluctuations with very low modulation rates. The maximum of this psychoacoustic quantity, fluctuation strength, is at modulation rates around *4 Hz*. The unit of psychoacoustic fluctuation strength is *vacil* (from vacillare, Latin: to fluctuate). The value *1 vacil* is achieved by a 1 kHz sine tone with a level of *60 dB* and a modulation rate of *4 Hz*.

Fourier transformation: The Fourier transformation is based on a mathematical theorem formulated by J. B. J. Fourier. This theorem states that every periodic signal form can be represented as a superposition of discrete periodic sine and cosine oscillations with different frequency und amplitude.

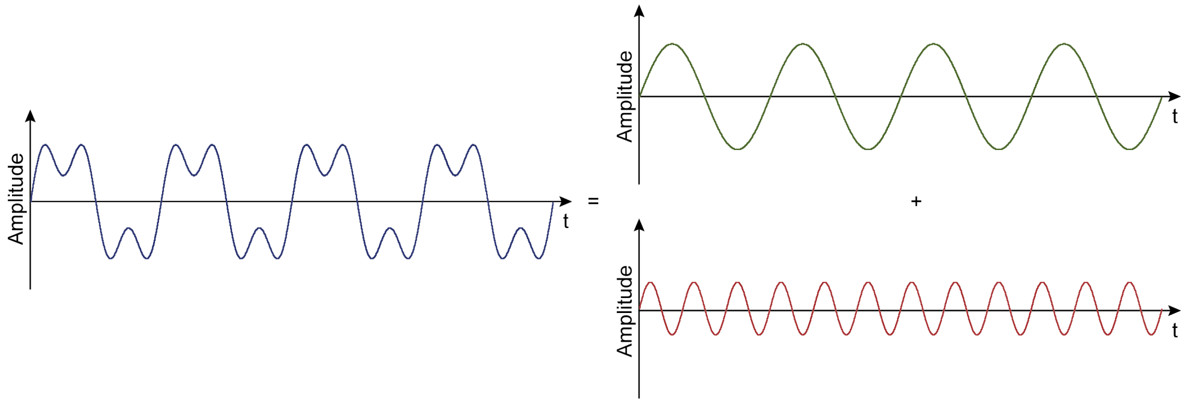


Figure 26: Periodic sum signal (blue) and contained frequency components (green and red)

The Fourier transformation is used to calculate the frequency spectrum of a signal, i.e., the signal is transformed from the time domain into the frequency domain.

DFT: An equidistantly sampled time-discrete time signal can be investigated by using the DFT (Discrete Fourier Transformation). The DFT maps a finite signal, which is continued periodically, onto a discrete periodic frequency spectrum.

FFT: The FFT (Fast Fourier Transformation) is an optimized algorithm for efficiently calculating the DFT. For calculating the FFT, the signal also has to be decomposed in respect to time before being transformed. For this purpose, the initial signal is divided into several blocks of N samples each (see also Leakage and Window function). With the FFT, the block length N has to correspond to a power of two, thus $N = 2^m$.

DFT and FFT are analyses with a constant bandwidth, e.g., the frequency nodes are equidistantly distributed on a linear frequency scale. On a logarithmic scale they are closer together at high frequencies.

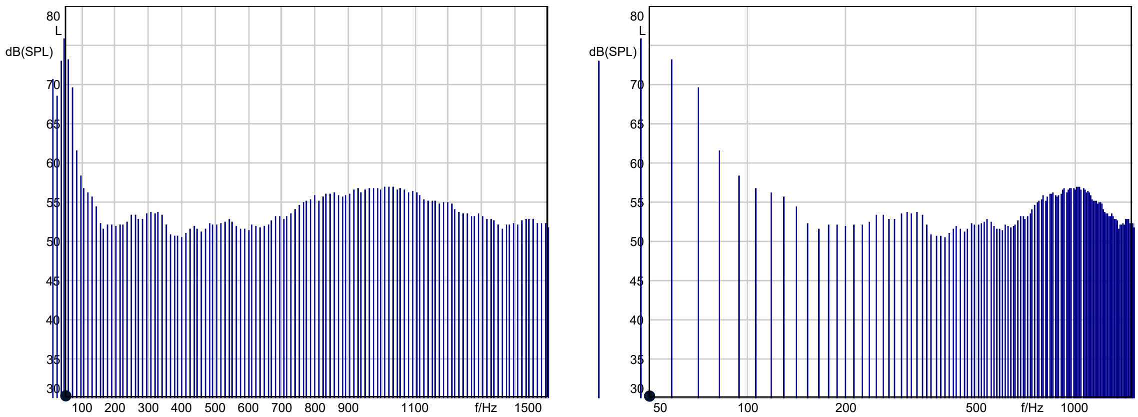


Figure 27: Frequency support points of an FFT (linear frequency axis (left), logarithmic frequency axis (right)

The frequency resolution Δf in Hz results from the sampling rate and the selected block length:

$$\Delta f = \frac{sampling\ rate}{block\ length}$$

Free field: see Sound field

Free field equalization: Free field equalization is a digital filter that can be used to modify a binaural recording to be compatible with a measurement microphone recording. However, a high degree of compatibility is only achieved if free field conditions were actually present during recording. For HEAD acoustics artificial heads, this means: sound incidence in free field, from the front at a distance of at least *3 m*, at ear canal height.

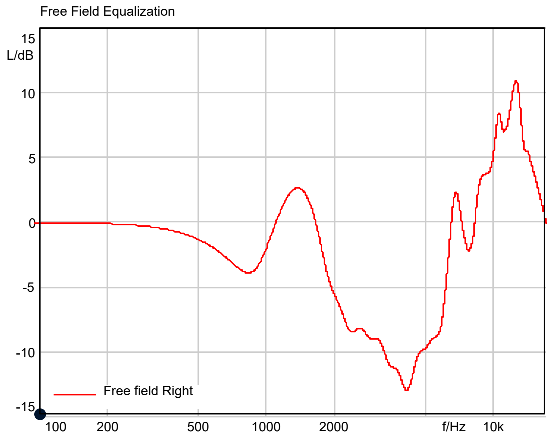


Figure 28: Free field equalization of an HMS IV artificial head measurement system (right ear)

Frequency: The frequency shows how many wavelengths the sound passes through in one second. The faster the vibration, the higher the frequency and the higher the perceived pitch. The frequency is measured in units of *Hertz* (*Hz*) and it is inversely proportional to the wavelength. For acoustic signals, the frequency and the wavelength are linked via the sound velocity:

$$Frequency = \frac{Sound\ velocity}{Wavelength}$$

The human ear can only perceive sound within a certain frequency range. The lowest audible frequency is about *20 Hz*. The highest audible frequency usually ranges between *16 kHz* and *20 kHz* and strongly depends on the listener's age. As the listener advances in age, the upper limit frequency of the audible frequencies decreases noticeably.

Frequency spectrum: The frequency spectrum, or simply the spectrum, indicates the sound pressure level of the different frequencies a sound consists of. The frequency spectrum of a time signal can be calculated by using the Fourier transformation.

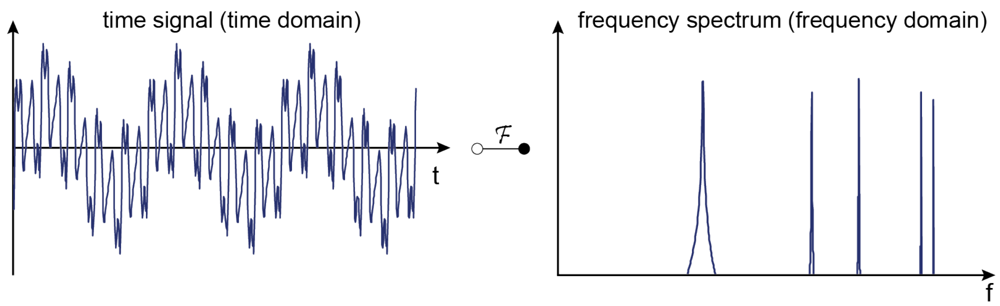


Figure 29: Time signal in the time domain and the Fourier transform in the frequency domain

Frequency weighting: see Sound pressure level

Ground loop: Ground loops occur when connected devices are grounded on different ground paths. A voltage difference in the ground potential causes a current flow which results in an interference voltage. In the AC mains voltage network, this interference voltage contains the network frequency (*50 Hz* or *60 Hz*) and a more or less high proportion of the network frequency harmonics. Ground loops can be avoided by connecting all devices to the same potential equalization path.

Group delay: The group delay is the delay of a signal's envelope when passing through a system, e.g., a filters. If the group delay is constant, the system has a linear phase response, and all frequencies pass through the system with the same delay. If the group delay is frequency-dependent instead, the system has nonlinear phase response, i.e., different frequencies pass through the systems at different speeds.

Group velocity: The group velocity is the velocity at which energy or information can be transmitted during sound propagation. In the case of a wave packet, i.e., a superposition of individual waves with different frequencies, the group velocity is the velocity at which the maximum of the envelope of a wave packet propagates. The individual waves of the wave packet each propagate with the phase velocity. The group velocity may differ strongly from the phase velocity.

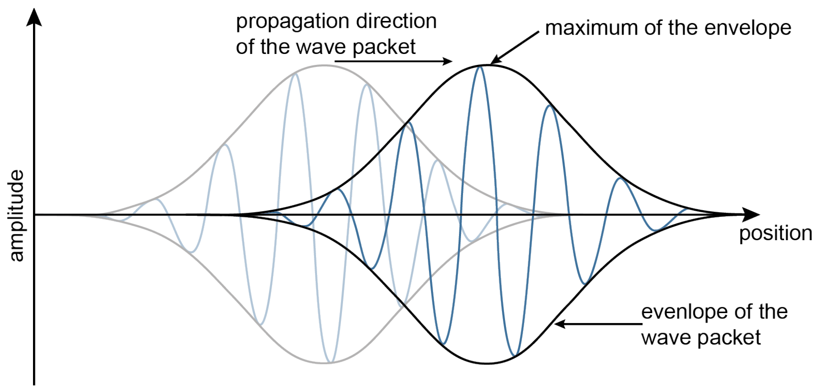


Figure 30: Group velocity of a wave packet

Harmonic: Harmonics are integer multiples of a fundamental frequency. Tones, such as those produced by musical instruments, are generally harmonically complex sounds, i.e., in addition to the fundamental tone, which is the tone with the lowest frequency, various integer multiples of the fundamental tone are audible as well.

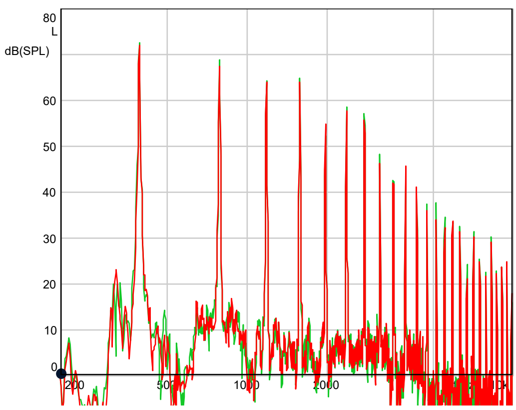


Figure 31: Averaged FFT of a sound with harmonics

HDF (HEAD acoustic Data Format): The HDF format is used to store HEAD acoustics data sets in a file. An HDF file can store multi-channel time signals as well as analysis results, i.e., a frequency spectrum. An HDF file always consists of a file header in plain text containing various descriptive entries. These include, for example, information on the type of the dataset stored, the number of channels, the channel names and the units. The file header is followed by the data set, i.e., the actual contents of the file, such as the time data. These data are not stored in plain text, and can only be read using appropriate software applications.
The file name extension normally used for HDF files is .hdf. However, other file name extensions that fulfill certain functions have been and are used for HDF files. In most cases, such files can be renamed in *.hdf and opened.

- .dat: These files contain time signal recordings and were created with HEAD NoiseBook, for example. However, there are also (very old) DAT files that do not represent an HDF format.
- .fft: These files contain frequency spectra generated by the software application ACQUA. In this format, for example, the individual equalization of the artificial heads of HEAD acoustics are generated and made available.
- .equ: These files also contain information on equalizations. The standard equalizations for the various binaural recorders in ArtemiS SUITE are provided as EQU files, for instance.

HEAD*link*: HEAD*link* is the specification of an interface for the electrical transmission of data, for the provision of a supply voltage and for synchronization. Usually, 8-pin LEMO connectors and corresponding cables are used for the transmission. Modules of the HEAD*lab* series are interconnected by HEAD*link*. Furthermore, SQuadriga II and SQuadriga III are equipped with HEAD*link* interfaces, so that they can be connected to each other and to HEAD*lab* systems.

Figure 32: HEADlink sockets

Head-related coordinate system: In the head-related coordinate system, the **median plane** is perpendicular to and interaurally equidistant along the ear canal axis, the connecting line between the two ears. The **horizontal plane** is parallel to the ground and perpendicular to the median plane. The deviation of a sound source in the horizontal plane is described by the horizontal angle of incidence φ (also referred to as azimuth angle). The horizontal angle of incidence is positive for clockwise deviation. An elevation of the sound source from the horizontal plane is described by the vertical angle of incidence δ (also referred to as elevation angle). The vertical angle of incidence is positive when the sound source is deviated upwards. The location of a sound source with respect to the head-related coordinate system is defined by the following specifications:

- Distance r
- Horizontal angle of incidence φ
- Vertical angle of incidence δ

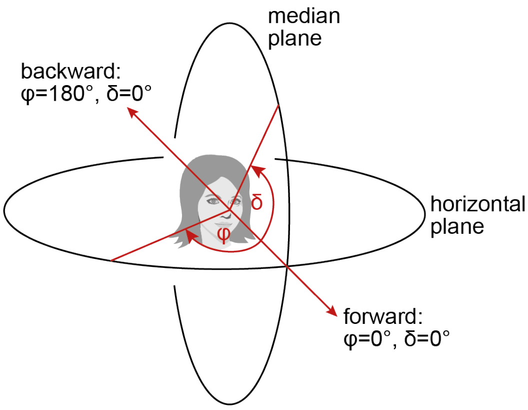


Figure 33: Head-related coordinate system

Head-related transfer function, HRTF: The head-related transfer function describes the transmission characteristics of the outer ear depending on the [frequency] and the location of the sound source. The head-related transfer function includes not only the transfer characteristics of the [auricle], it also captures the acoustically effective influence of the entire external geometry of a test person, consisting of the upper body, shoulders, head and outer ear. The head-related transfer function is very different for each individual due to the respective dimensions of the outer geometry and the ear canal.
The **monaural head related transfer function** is the ratio of the sound pressure at any sound incidence direction to the sound pressure at sound incidence from a reference sound incidence direction. As a rule, a plane sound wave from the front is taken as the reference sound (horizontal sound incidence angle $\varphi = 0\,°$, vertical sound incidence angle $\delta = 0\,°$).

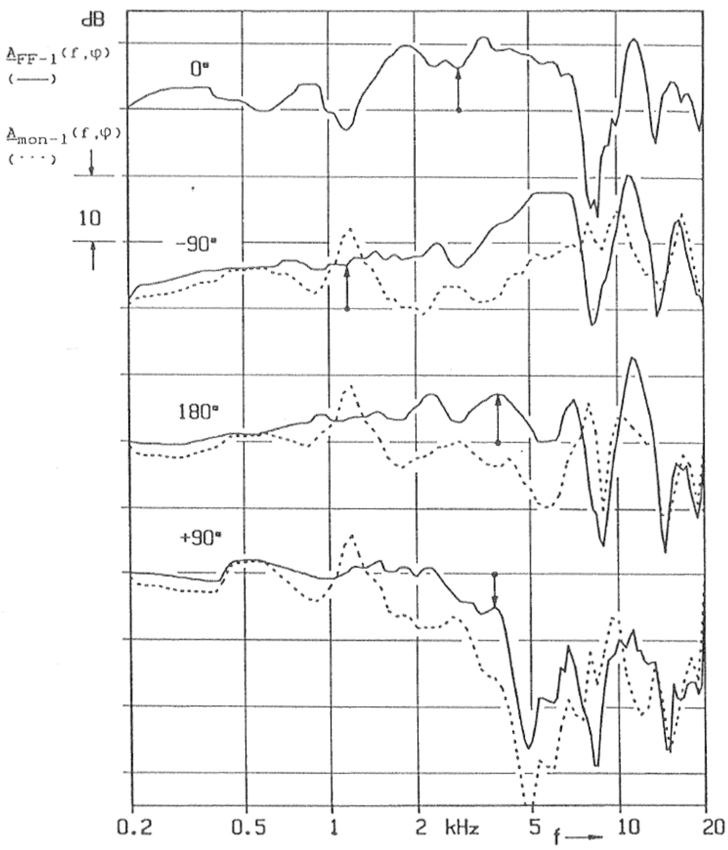


Figure 34: Measured outer ear transfer function of a test person's left ear for the four main directions of sound incidence in the horizontal plane (solid line: free field; dashed line: monaural transfer function); (from Genuit: A Model for the Description of Outer Ear Transfer Properties, doctoral thesis, 1984)

Headroom: The headroom is a range between the nominal level and the highest level that can be processed without distortion. The headroom is used to record any level peaks that may occur without overload. In devices from HEAD acoustics, the headroom is *6 dB*.

Helix: see Auricle

Hertz: Hertz (Hz) is the unit for specifying the frequency, i.e., the number of repetitive processes per second. The following applies: $1Hz = \frac{1}{s}$

Horizontal plane: see Head-related coordinate system

HRTF: see Head related transfer function

ICP® (integrated circuit piezoelectric): ICP® sensors feature built-in piezoelectric electronics with an impedance converter. The impedance converter transforms the measurement signal into a signal with lower impedance which can be transmitted over long distances with low loss, even without low-noise special cables. To operate, ICP® sensors require a constant direct current which must be provided by the measuring device. The supply current and the sensor signal can be transmitted together via a simple coaxial cable.

ID equalization: ID equalization is a filter, typically implemented digitally, for equalization of binaural recordings.
Based on a systems-theory description of the outer ear, ID equalization involves only the correction of the direction-independent parameters, i.e., the resonances generated by the cavum conchae cavity and the ear canal input. Since this equalization corrects only the direction-independent changes, it is called ID (Independent of Direction) equalization. ID equalization is used for all sound field situations in which neither free field- nor diffuse field conditions prevail. In general practice, the ID equalization of a binaural transducer is far more frequently the approprate choice than either FF or DF equalizations.

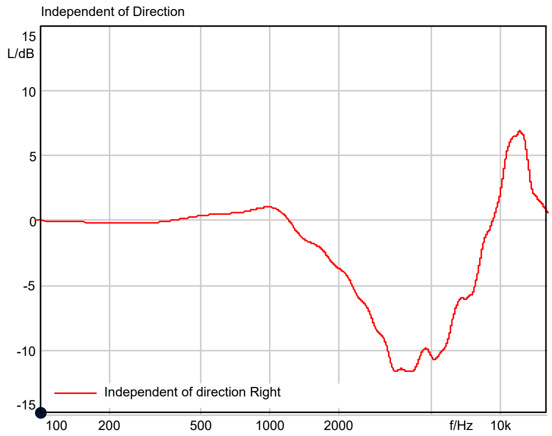


Figure 35: ID equalization for an artificial head measurement system HMS IV (right ear)

Impedance: Impedance is a measure of the resistance to the propagation of waves in a given medium. The greater the difference in impedance between two media, the more energy is reflected when a wave passes from one medium to the other.

Impact hammer: An impact hammer is a hammer equipped with a force sensor, used to measure the force of the applied impulse. This type of hammer can be used to determine the acoustic properties of a structure. To do this, the impulse hammer is used to excite the structure to vibrate and accelerometers and microphones are used to measure the structure-borne noise or the radiated airborne noise. By simultaneously measuring the force introduced with the hammer and the induced accelerations, or sound pressures at the different measuring points, the transmission behavior of the component can be determined.

Figure 36: Impact hammer

Interaural differences: Interaural differences are differences between the signals of the listener's left and right ear:

Interaural level difference, ILD: These are differences in sound pressure level between the two ear signals caused by diffraction, reflection and shading.

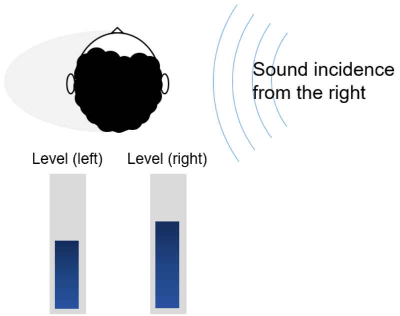


Figure 37: Interaural level differences

Interaural time differences, ITD: Interaural time differences characterize the delay between the two ear signals at the time of arrival of sound waves. The farther the sound source direction deviates from the median plane, the greater the difference in time of arrival.

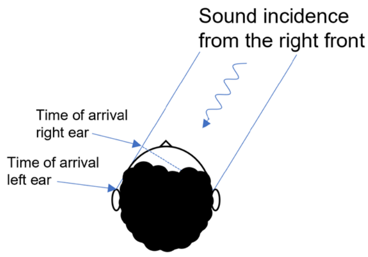


Figure 38: Interaural time differences

Interference of sound waves: If sound waves overlap, interference can occur, and the amplitudes of the sound waves will add. The amplitude of the resulting wave depends on the phase difference of the interfering sound waves. If a wave crest meets a wave trough, destructive interference occurs, i.e., the wave is cancelled. If a wave crest meets a wave crest, constructive interference occurs, i.e., the wave is amplified.

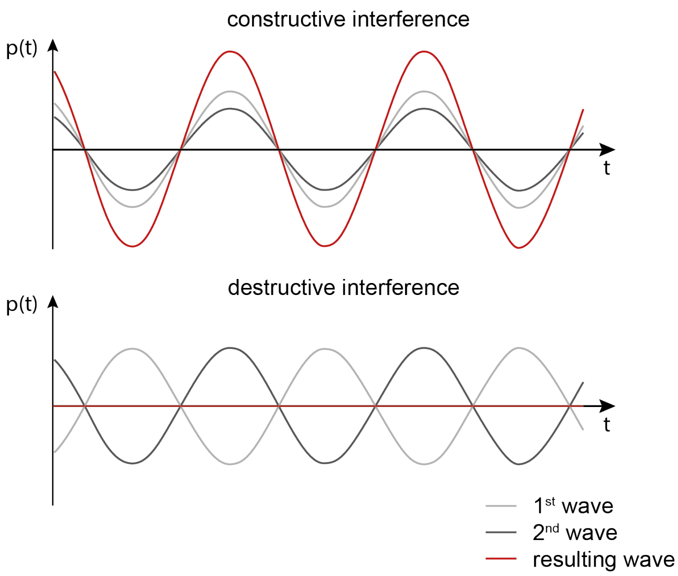


Figure 39: Constructive and destructive interference

Jack plug: The jack plug is often used for connecting headphones. The jack socket (usually simply called jack) contains contact springs which simultaneously serve to transmit the audio signal and to mechanically secure the plug connection. Depending on the number of poles, the jack plug has one (sometimes two) extension rings in addition to the tip and the shaft (called sleeve). Jack plugs are available with various shaft diameters: the usual diameters are 3.5 mm and 6.35 mm. Jack plugs with a shaft of 3.5 mm diameter are also called „mini plugs“. Jack plugs with a shaft of 6.35 mm diameter are also called „large jack plugs“ or simply “phone plugs”.

Figure 40: Jack plug

Jitter: Jitter is an unwanted fluctuation of a digital signal. Jitter can have various causes and effects. Two examples are explained below.

- Fluctuations in the transmission clock may lead to fluctuations in the sampling of a signal (i.e., jitter of the sampling rate). If the sampling timing fluctuates, an amplitude value is displayed at a time at which it was not present in the original. This can distort the sound of the recordings.
- Jitter effects can also be caused by too low a sampling rate. The following figure shows the sampling of a pulse signal at two different sampling rates. The sampling rate in the upper figure is higher than the one in the lower figure. To determine the time Δt between two pulses, the sampling points between two consecutive 0→1 jumps are counted and divided by the respective sampling rate.

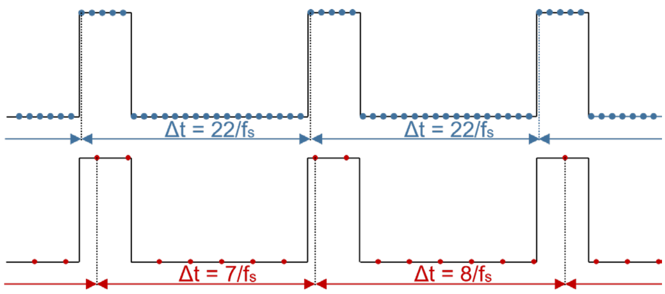


Figure 41: Pulse signal sampled at different sampling rates

In the lower figure, Δt shows once $7/f_s$ and once $8/f_s$, although the distance between the rising edges is equidistant. This means that the calculated momentary value for Δt jumps back and forth between two or more values. This fluctuation caused by too low a sampling rate is also called jitter.

Jury test: see Listening test

Leakage: The leakage effect occurs when the block length of the FFT is not an integer multiple of the signal under investigation. With the FFT, the time signal has been divided into blocks. When analyzing these blocks, a periodic continuation of the time signal is implied, i.e., it is assumed that the signal can be repeated infinitely often by concatenating the data block together. In reality, however, this is rarely the case; as a matter of fact, the signal level at the beginning is usually different from the signal at the end. Thus, concatenating the data block causes discontinuities, which lead to a widening of the spectral lines in the resulting spectrum. The outflow of signal energy to adjacent frequencies of the original frequency is referred to as leakage. Thus, the spectrum resulting from the FFT calculation is not identical with the spectrum of the original signal, but a blurred version.

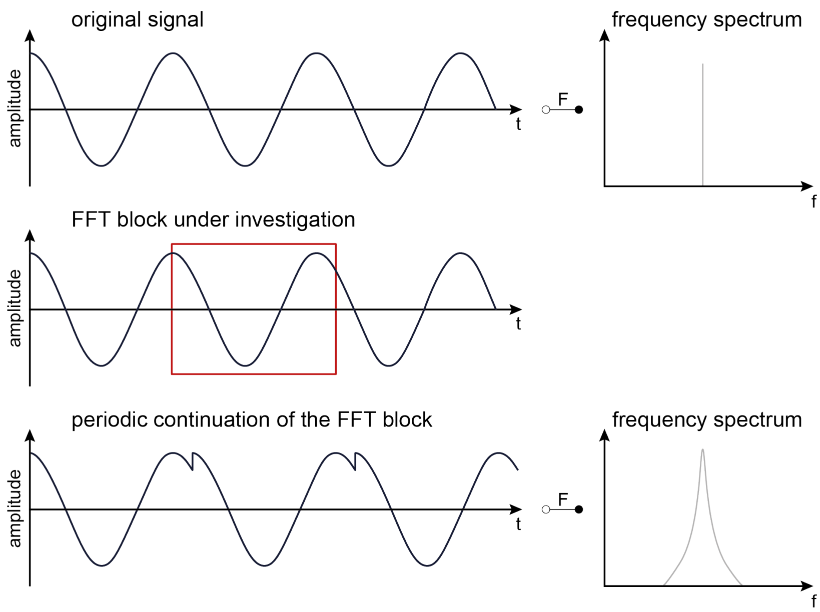


Figure 42: Leakage effect

However, this effect can be reduced by using a suitable window function.

LEMO: LEMO is a manufacturer of specific connectors and corresponding cables. LEMO connectors are equipped with a connection system featuring a self-locking mechanism that prevents accidental disconnection. LEMO connectors and cables are available in different sizes and with different characteristics. Both data and voltages can be transmitted via a LEMO cable and appropriate connectors.

Listening test (or jury test): Listening tests are used to investigate the perception of sound events under defined conditions. Within the scope of a listening test, the acoustic perception of sound events is investigated under defined conditions. The aim is to establish a relationship between the sound stimulus and the auditory event. Listening tests form the basis for psychoacoustic investigations. In addition, listening tests are an important tool to check and improve the sound quality of products. Listening tests can be used to check if the sounds emitted by a product are accepted by later customers or if the customers‘ expectations are met. In order to obtain valid results from a listening test, it has to be carefully planned and evaluated.

Figure 43: Listening test with several participants

Loudness: The psychoacoustic loudness is the sensory quantity of the human perception of volume. This parameter is used to map the human loudness perception of acoustic signals in a linear scale, so that a sound that is perceived as twice as loud has a loudness value that is twice as high. The unit of loudness is *sone* (from sonare, Latin: to sound). A sine tone of frequency *1 kHz* with a sound pressure level of *40 dB* has by definition a loudness of *1 sone*. The specific loudness shows the distribution of loudness over critical bands.
Loudness is defined in various standards (DIN 45631/A1, ISO 532-1, ISO 532-2, ISO 532-3 and ANSI S3.4 2007).

Loudness level: The volume perception of the human ear is frequency-dependent. The sensitivity of the ear decreases at high or low frequencies. Therefore, sound events with the same sound pressure level but different frequencies do not always cause the same loudness perception in humans. The loudness level with the unit *phon* is a comparative unit indicating what sound pressure level in *dB(SPL)* a 1 kHz sine tone would need to have in order to evoke a comparable loudness perception as the sound event to be evaluated. For a sinusoidal tone with a frequency of *1 kHz*, the sound pressure level in *dB* always corresponds to the volume level in *phon*. Sinusoidal tones with other frequencies, and other sounds, require other sound pressure levels to evoke the same volume impression as a 1 kHz sinusoidal tone.

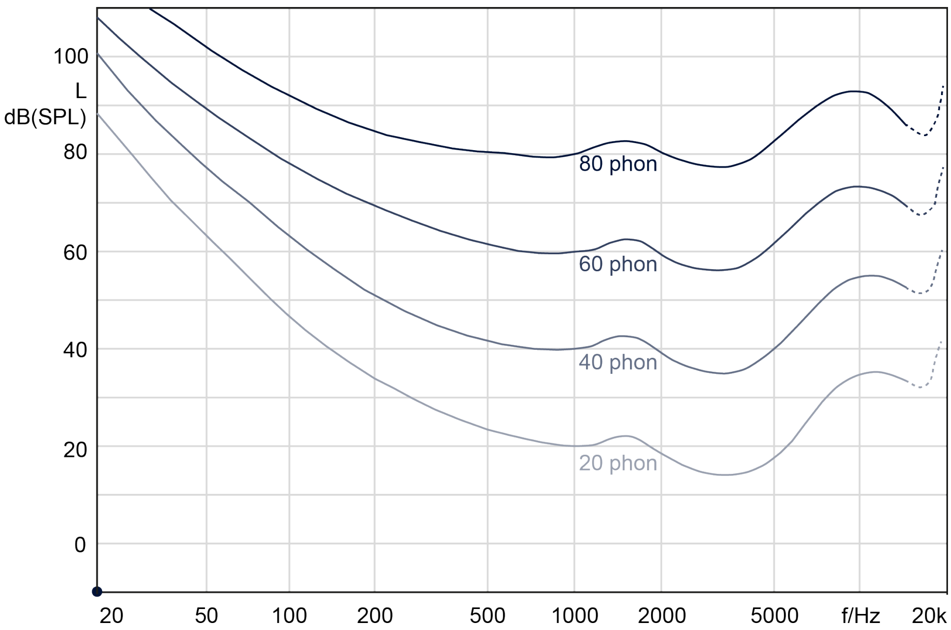


Figure 44: Curves of comparable loudness perception according to ISO 226:2003

Examples: A 1 kHz sine tone with *80 dB* sound pressure level has a volume level of *80 phon*. A 50 Hz sine tone that is perceived to be as loud as a 1 kHz sine tone with *80 dB* has to be played back with a significantly higher sound pressure level (more than 100 dB); only then will the *50 Hz* tone be perceived to be as loud as the 1 kHz sine tone with *80 dB*. The 50 Hz sine tone raised in level has a volume level of *80 phon*.

Masking: Masking means that a present sound cannot be perceived by the human ear, or can only be perceived very softly because it is masked by another sound.
With spectral masking, simultaneously occurring noise components lying within a critical band influence each other. The part with the highest level makes the parts with similar frequency but significantly lower level inaudible. The effect of spectral masking can be reduced by increasing the frequency spacing.
Temporal masking masks a quieter sound that occurs just before or after a very loud sound. Post-masking extends over a much longer period of time than pre-masking. The exact duration of the pre- and post-masking depends on the sound pressure level and the duration of the masking.

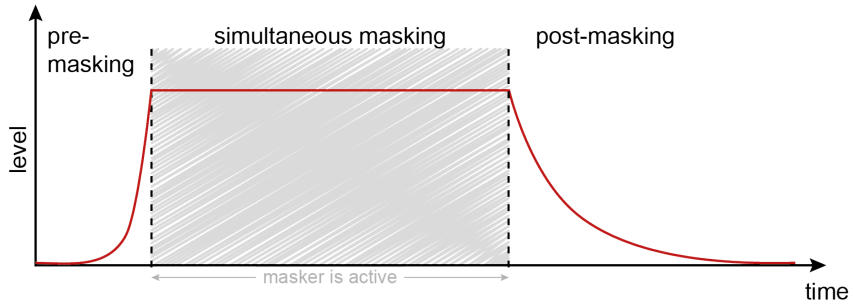


Figure 45: Temporal masking

Acoustic masking can have negative consequences, as sounds that contain important information become masked and become inaudible. On the other hand, a lack of masking can make unwanted sounds audible. This can be the case, for example, with electrically driven vehicles, where the masking sound of the combustion engine is eliminated. As a result, noise problems from ancillary components, for example, become audible that were not noticeable in vehicles with combustion engines.

Median plane: see Head-related coordinate system

Microphone: Microphones are sensors, that convert the sound pressure fluctuations of the air into electrical signals. There are microphones with different types of transducers available on the market. The most common ones used in measurement technology are condenser and electret microphones.

Condenser microphone: Condenser microphones have a thin diaphragm which is electrically insulated and mounted in front of a metal plate (back plate), with an air gap between. This setup corresponds to a plate capacitor with a defined electrical capacitance. Sound waves cause the diaphragm to vibrate, changing the distance between the diaphragm and the metal plate and thus the capacitance of the capacitor. If the capacitor is charged with a bias voltage via a high-impedance resistor, these capacitance fluctuations can be converted into an electrical voltage signal. The bias voltage is also called polarization voltage. In order to operate a condenser microphone, a measuring instrument is required that can provide an appropriate polarization voltage.

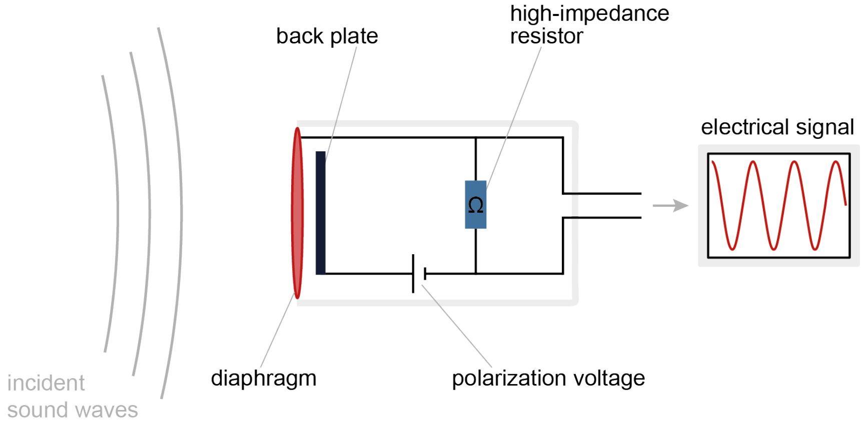


Figure 46: Schematic setup of a condenser microphone

Electret microphone: The electret microphone is a special condenser microphone with an electret layer applied to the counter electrode. This layer has a constant electrical charge which provides the voltage for polarization. An electret microphone therefore does not require an externally supplied polarization voltage.

MEMS microphone: MEMS microphones (Micro Electro Mechanical System) are miniaturized microphones for direct use on electronic boards. The main advantages of MEMS microphones are their small dimensions, low power consumption and cost-effective production. MEMS microphones are therefore used, for example, in mobile phones. MEMS microphones are therefore used in conjunction with mobile phones or in microphone arrays, for example.

Microphone array: A microphone array consists of a number of microphones arranged in a defined pattern. They are available as two-dimensional arrays, such as spiral arrays and ring arrays or arrays with randomly distributed microphones. Even three-dimensional arrays are possible, e.g., spherical arrays.

Microphone arrays can be used for beamforming. The size and structure of an array have a direct influence on the frequency range with beamforming. The larger the array (distance between the microphones), the lower the lower cut-off frequency, and the smaller the distance between the microphones, the higher the upper cut-off frequency.

Figure 47: Spiral array

Modal analysis: Experimental modal analysis is a method for analyzing the dynamic properties of linear, time-invariant structures. For this purpose, modal parameters (natural frequency and natural form and damping) are determined which, in turn, can be calculated from measured transfer functions using suitable software applications. To determine the transfer functions, the structure under investigation is excited, for example, with an impact hammer or a shaker. The time signals of the exciting force and the response of the system to the applied force are measured simultaneously and evaluated afterwards.

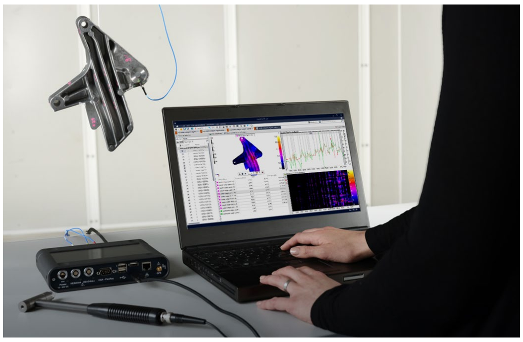


Figure 48: Modal analysis on a torque arm

Modulation: If a carrier signal is varied by another signal, it is referred to as a modulated signal. The modulation rate is lower than the carrier frequency. A signal can be amplitude-modulated or frequency-modulated. An amplitude-modulated signal (see figure) changes its amplitude depending on the modulation frequency. The degree of modulation determines how strongly the amplitude of the modulated carrier signal is affected, that is, how significant the magnitude changes are.

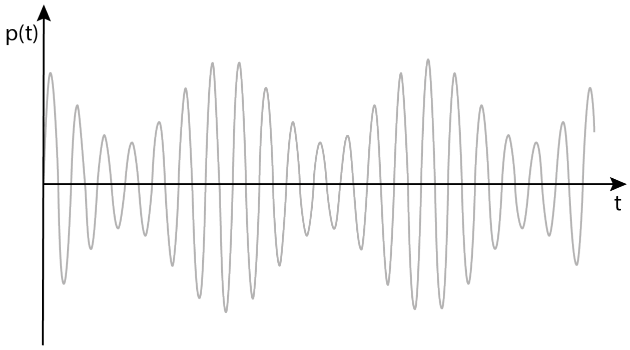


Figure 49: Amplitude-modulated sine signal

A frequency-modulated signal changes its pitch depending on the modulation rate. The frequency deviation, i.e., the difference between the maximum and minimum of the instantaneous frequency, determines how strongly the frequency of the carrier signal is affected. Examples for frequency-modulated signals are the vibrato of string instruments and the American police siren.

At low modulation frequencies rates, when changes are slow, the human ear is able to follow the change and the sound is perceived as fluctuating (→ fluctuation strength). If the modulation rate rises, the human ear is no longer able to follow the individual changes and the signal is perceived as rough (→ roughness). If the modulation rate is yet further increased, the hearing system will eventually not be able to recognize the modulation as roughness and the sound will be perceived as steady, perhaps with timbre changes.

Monaural: hearing with one ear

MP3: MP3 is a method for lossy compression of audio data. During conversion, only the signal components that are actually perceptible to humans are stored. Signal components that are masked by other components, for example, and are not audible, are not stored. This results in considerable data reduction with a hardly reduced perceived audio quality. Data reduction offers many advantages in storing and transferring audio files (less storage space and lower data rate). Despite the fact that more advanced methods for data reduction of audio signals are available nowadays, MP3 is still very common. Audio files compressed by the MP3 method have the file name extension .mp3.

Natural frequency: The natural frequency is a property of an oscillatory system. It is the frequency at which the system vibrates after a single excitation. A system with several degrees of freedom can have several natural frequencies. Each natural frequency has an associated eigenmode. This eigenmode is a specific vibration pattern that the component shows when vibrating at this frequency.

Noise: One definition of noise is a sound that consists of a superposition of many sound waves with different amplitudes and frequencies. Another definition of noise is unwanted or undesirable sound. Noise occurring during the measurement or transmission of sound is undesirable in most cases and is considered to be disturbing noise. There are, however, applications in which noise signals are specifically used (e.g., when determining transfer functions).

White noise: White noise is generated during stochastic processes. It has a frequency-independent spectral power density, i.e., the level is constant over the entire frequency range. Despite the evenly distributed energy, white noise is perceived by humans as a noise with a strong emphasis on high frequencies.
In an analysis with consistently distributed frequency support points (e.g., FFT), the curve level shows a frequency-independent progression. In an analysis whose bandwidth in Hz increases towards high frequencies, (e.g., 1/3-octave spectrum), the curve level increases towards high frequencies.

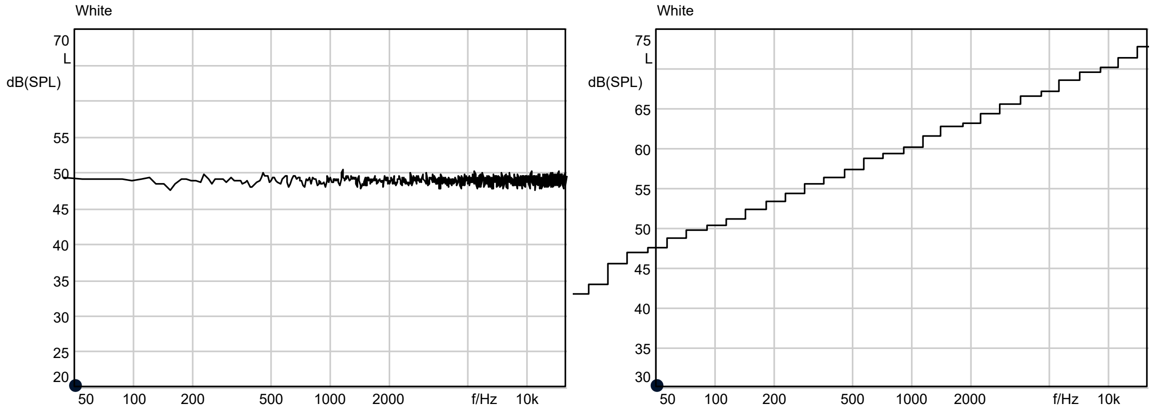


Figure 50: White noise; left: FFT analysis, right: 1/3-octave level analysis

Pseudo white noise: The spectrum of white noise can be approximated with pseudo-random noise. Noise generated by a computer using deterministic random generators is not really random. It is referred to as pseudo-random noise because it is generated from deterministic regularities.

Pink noise: Pink noise is noise whose amplitude decreases at *3 dB* per octave as the frequency increases, in other words pink noise has equal energy per constant-percentage bandwidth. In an analysis with consistently distributed frequency support points (e.g., FFT), the curve therefore tends to fall off towards high frequencies. An analysis that does not have a constant absolute bandwidth but a constant relative bandwidth (e.g., 1/3-octave spectrum) shows a frequency-independent progression (same band level regardless of the frequency band). The drop in pink noise power towards high frequencies is compensated for by the widening filters (in Hz) of the third-octave level analysis.
Pink noise is perceived by humans as a sound in which all frequency ranges are perceived to be almost equally loud.

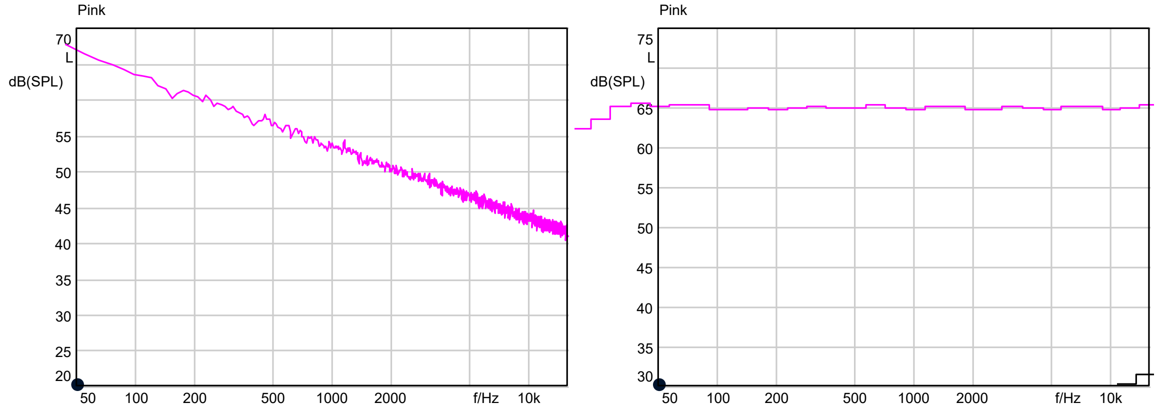


Figure 51: Pink noise; left: FFT analysis, right: 1/3-octave level analysis

Uniform masking noise: The uniform masking noise is a broadband noise that generates a frequency-independent masked threshold for people with normal hearing. This noise is used, for example, for psychoacoustic listening tests to investigate masking effects.

Noise floor: The noise floor is the sum of all unwanted signals that occur during a measurement. Every electroacoustic system emits more or less strong, self-generated noise that superimposes upon the useful signal. This interfering noise can either be caused by the mains frequency or by the components themselves.

NVH: The abbreviation NVH stands for Noise, Vibration and Harshness. NVH is a collective term for audible or tangible vibrations. “Noise“ stands for the audible airborne noise, “vibration“ for the perceptible structure-borne noise and “harshness“ for the transition range between *20 Hz* and *100 Hz* in which the human being can both hear and feel the vibrations. In general, the NVH properties of a product influence the customer in his evaluation of the overall product quality. NVH measures are those measures that are intended to quantify and improve acoustic comfort.

OBD-2: OBD means "On Board Diagnostics" and is a vehicle diagnostic system. It was introduced in 1988 by the California Air Resources Board and is used for continuous monitoring of all exhaust gas influencing systems in vehicles. Any faults that occur are indicated to the driver by a control lamp and can be read out via standardized interfaces. Since the transmitted variables are subject to standardization, they are coded independently of the manufacturer. After its introduction, the OBD system was further developed, the second development stage is referred to as OBD-2. The interface is a 16-pole socket which needs to be accessible within a radius of 1 m around the driver's seat. The data provided on the socket can be read out via special OBD-2 cables. This includes, for example, the current engine speed and the vehicle speed.

Octave analysis: An **octave** is a frequency range whose upper cut-off frequency is twice the lower cut-off frequency. In order to calculate an octave spectrum, the signal to be analyzed is first split by a digital filter bank of octave filters which are connected in parallel. The level of each partial signal is then determined and displayed as a function of the center frequency of the respective octave filter.

1/3-octave analysis: 1/3-octave filters are used to determine a 1/3-octave spectrum. These filters subdivide the octave bands again into three parts.

Octave and 1/3-octave filters do not have a constant absolute bandwidth, but a constant relative bandwidth, i.e., the center frequencies of the filters are equidistantly distributed on a logarithmic frequency scale. 1/3-octave filters are particularly well suited to reproduce human perception because the width of 1/3-octave filters above 500 Hz corresponds approximately to the width of critical bands.

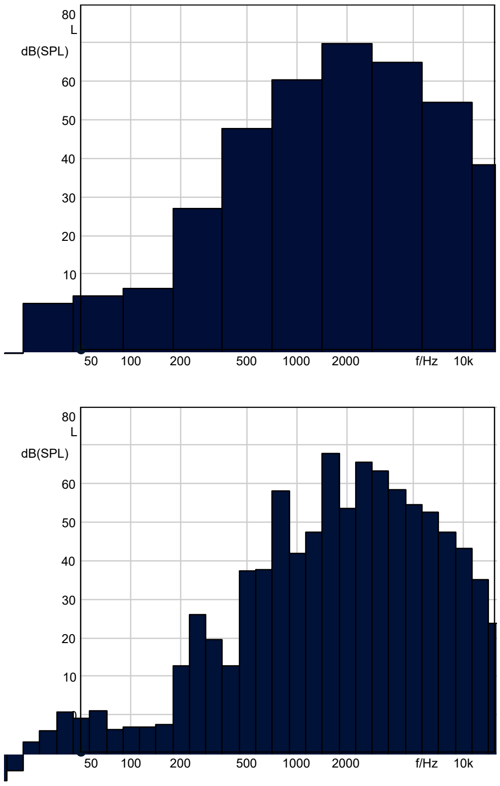


Figure 52: Octave analysis (top) and 1/3-octave analysis (below) of the same time signal

Operating deflection shape analysis, ODS analysis: The operating deflection shape analysis analyses the vibration behavior of a structure in a certain operating state. To determine the operational deflection shapes, the vibrations during operation are measured at different points. For these measurements, it is essential to record not only the magnitude of the amplitudes at each measuring point, but also the phase relationship of the measured signals to each other. The results of the operating deflection shape analysis can, for example, be displayed in an animated form by means of appropriate software. This form of visualization facilitates the interpretation of the operating deflection shapes.

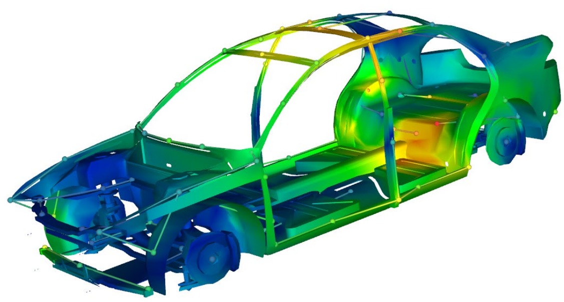


Figure 53: ODS visualization on 3D model

Order analysis: The speed of the motor plays the most important role in the development of order-related sounds: certain sound emissions generated according to the angle of rotation repeating after each revolution, so that the periodic oscillations they cause correspond in frequency to the frequency of rotation of the motor, or multiples thereof. Frequencies corresponding to the RPM or its multiples are called orders. The first order is identical to the frequency of the motor speed (i.e., a first-order event occurs once per revolution); the second order corresponds to the frequency of the first order multiplied by a factor of 2, and so on. In the order analysis, the level of these orders is calculated.

An RPM-dependent FFT analysis (left) shows the level curve of a sound file as a function of speed (x-axis) as well as of frequency (y-axis). Such a spectrum thus allows the user to read the level of a given speed and frequency. An RPM-dependent order spectrum (right) also shows the speed on the x-axis. However, the y-axis does not show the frequency in *Hz*, but the rotational speed frequency and its multiples, i.e., the orders. The frequency axis is distorted according to the current speed, so that the orders are no longer displayed as curves, but as straight lines in the diagram. The diagram thus shows the level as a function of the speed and the order.

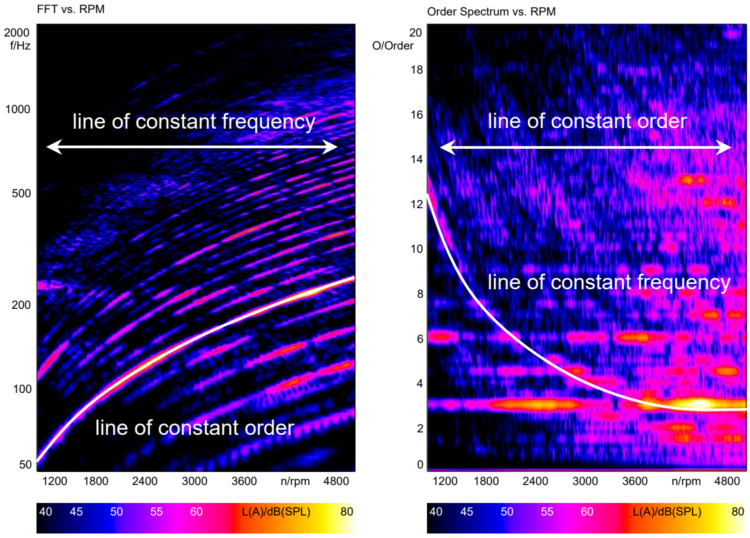


Figure 54: Comparison FFT and order analysis

Sounds from three-phase electric motors often result in an order spectrum that contains not only the usual orders proportional to the speed, but also other frequencies. These include the inverter running frequency and sidebands surrounding that frequency. The frequencies of these sidebands are determined by the rotating field frequency of the three-phase motor. Since the rotating field frequency changes in proportion to the motor speed, the fan-shaped pattern shown here forms with rising and falling frequencies, i.e., positive and negative orders relative to the inverter running frequency, as the RPM increases (left panel). For a simplified evaluation of the order levels of these sound components, ArtemiS SUITE offers the use of a frequency offset in the order calculation (right panel).

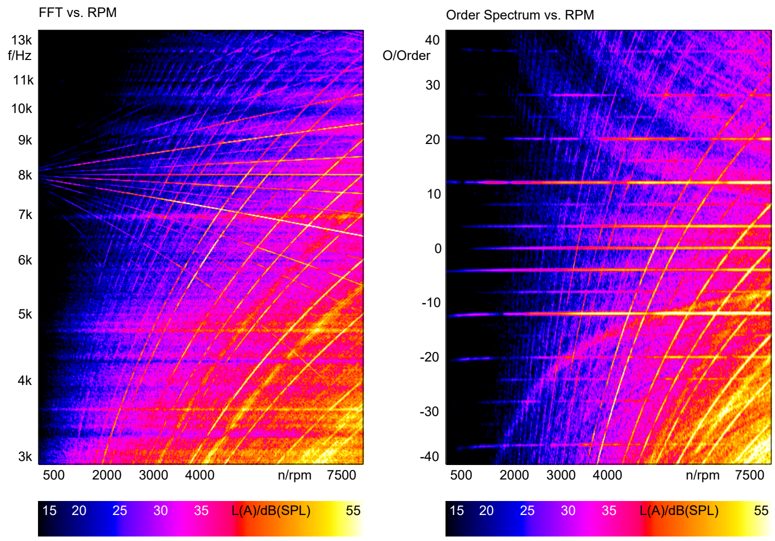


Figure 55: Order fan of an electric drive (left); order spectrum calculated with frequency offset of 8kHz (right)

Overload (or overdrive): If an input signal contains values that are beyond the dynamic range of the recording device, the signal will be overloaded. Usually, values that exceed the possible value range in a digital system are reduced to the highest possible value. The curve is thus clipped.

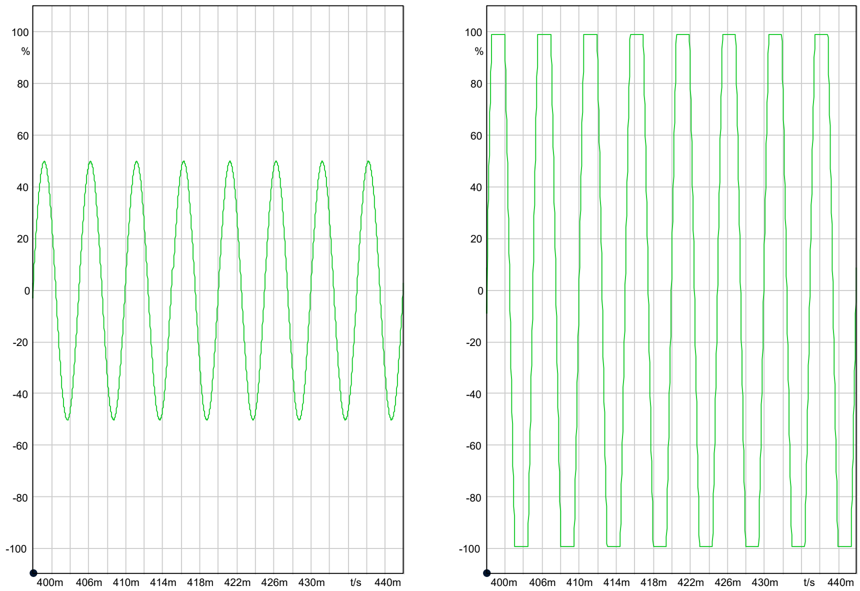


Figure 56: Correctly controlled recording of a sinus tone (left), overloaded recording of a sinus tone (right)

The recorded signal is distorted and no longer corresponds to the original input signal.

Oversampling: Generally, oversampling refers to the sampling of a time signal at a higher sampling rate than required by the sampling theorem. When acquiring speed information in the form of pulse channels, oversampling means that the pulse channels are sampled at a higher sampling rate than the audio signals recorded in parallel.

Phase: Phase is the oscillation state of a periodic signal. The phase is specified in angular degrees. The **phase position** refers to the relationship between two periodic oscillations: Two oscillations with the same frequency but not simultaneously occurring zero crossings have a phase position shifted against each other.

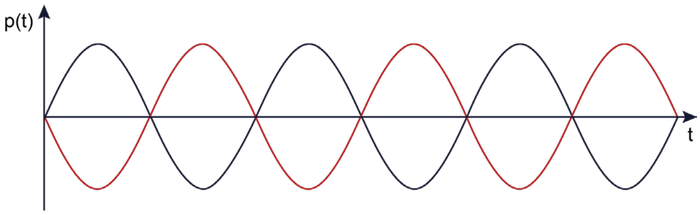


Figure 57: Two sine oscillations with 180° shifted phase

The phase velocity is the speed with which a particular oscillation phase propagates; it corresponds to the speed with which an excitation propagates in the carrier medium.

Pitch: In physical acoustics, the pitch is equated with the frequency of an audible tone. In the case of a sound with a fundamental tone and differently pronounced harmonics, the frequency of the fundamental tone corresponds to the pitch. The perceived, psychoacoustic pitch of sine tones is expressed in the unit *mel*.

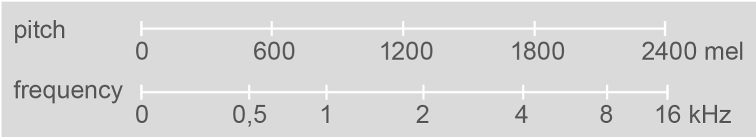


Figure 58: Perceived pitch and frequency according to Zwicker

A sound with a high frequency is perceived as a high-pitched tone; a sound with a low frequency is perceived as a low-pitched tone. The psychoacoustic pitch is approximately proportional to the critical band classification in Bark.

Psychoacoustics: Psychoacoustics studies the relationship between the physical quantities of a sound event and the auditory event evoked. For this purpose, physical parameters such as sound pressure level, frequency and degree of modulation are mapped to aurally-appropriate parameters. In contrast to the physical quantities, these psychoacoustic parameters are intended to map linearly the human perception. This means that a doubling of the psychoacoustic parameter magnitude corresponds to a doubling of the corresponding auditory perception. This makes it easier to evaluate the results of the analysis in terms of their effect on the human auditory system. The psychoacoustic parameters include loudness, sharpness, roughness, fluctuation strength and tonality.

Quantization: Together with sampling, quantization serves to digitize an analog signal. By sampling the analog signal at a defined equidistant sampling rate, the conversion to a time-discrete signal is performed; by quantization with a defined equidistant resolution, the conversion to a value-discrete signal is performed. Differences between the original analog signal and the resulting digital signal caused by quantization generate unwanted interference signals. These are called quantization noise. The quantization noise can be reduced by a high sampling rate and a high resolution.

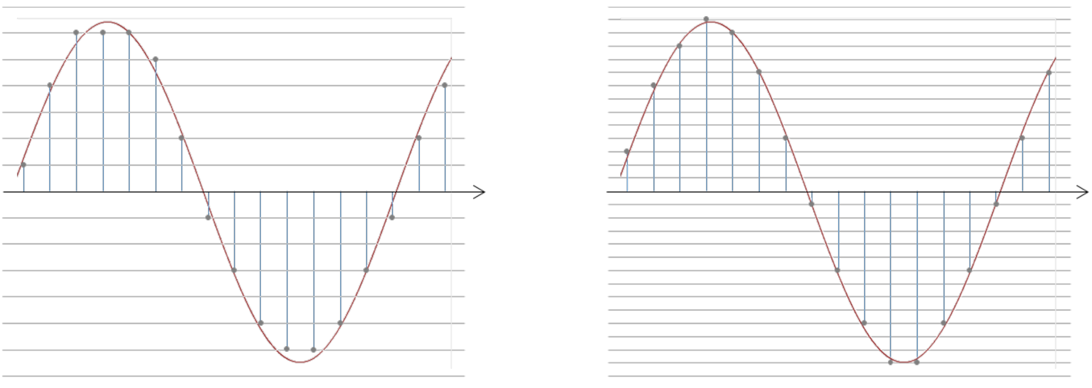


Figure 59: Quantization of an analog signal with lower resolution (left) and higher resolution (right)

Range: The range of a recording device determines the magnitude range in which the values transmitted by the sensor are recorded correctly. If several ranges are available, a suitable one has to be selected:

- **As large as necessary:** The range has to be as large as necessary, so that the expected measuring signal as well as any level peaks that may occur can be recorded without distortion.
- **As small as possible:** On the other hand, the range is to be selected as small as possible in order to obtain a measuring signal with a high tone-to-noise ratio. Operationally, the recordist needs to select the lowest numerical range in which the signal avoids overload, i.e., has the highest recording level without overload.

Reflection of sound waves: Sound waves propagate from the sound source until they impinge upon an obstacle, such as a wall. Depending on the nature of this obstacle, some of the sound energy is reflected back from the obstacle, similar to the reflection of a beam of light hitting a mirror. A wave's angle of incidence is always equal to its angle of reflection.

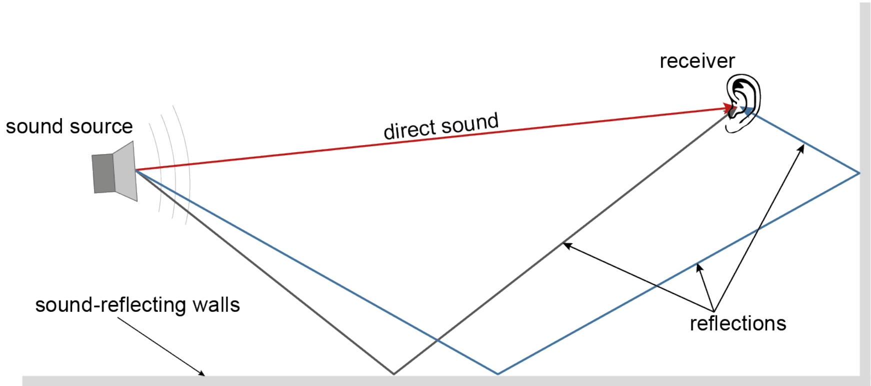


Figure 60: Reflection of sound waves

Resolution (or bit depth): For digital systems, the resolution indicates how many different values a sample can have. It thus determines in how many gradations the amplitude of a signal is represented (see also quantization). The higher the resolution, the more precisely the amplitude of the analog signal can be described. The resolution is specified in bits. With a resolution of 16 bits, a sample can have one of 2^{16} amplitude values.

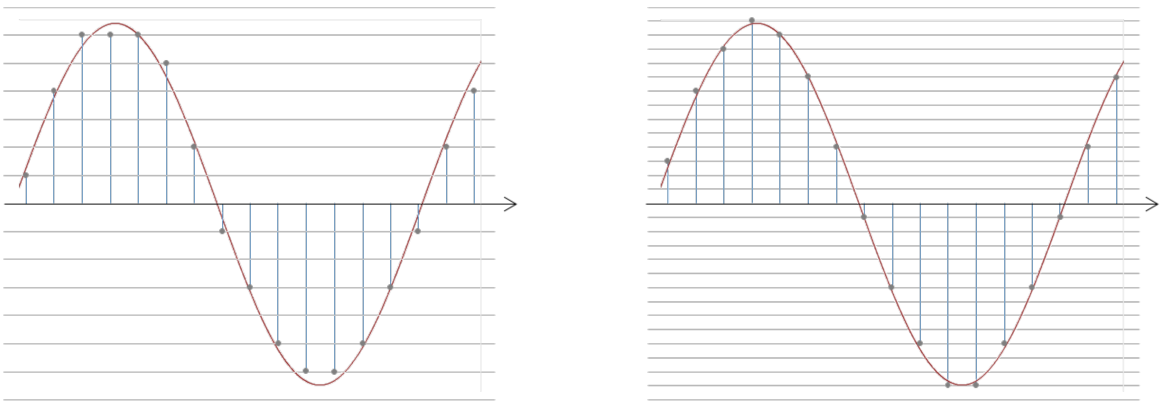


Figure 61: Digitizing with different resolutions: lower resolution (left), higher resolution (right)

Resonance: If an oscillating system is excited close to its natural frequency or an integer multiple thereof, the amplitude of the forced oscillation reaches a maximum. This is called resonance. Resonance occurs, for example, when sound in the frequency range of *2 kHz* to *4 kHz* strikes the human ear. The self-amplification of the incident sound by the ear canal can be more than 10 *dB*.

Reverberation: Reverberation occurs as a result of continuous sound reflections in a limited space. In rooms with little absorbing surfaces, such as churches or indoor swimming pools, reverberation is particularly high. In contrast, no reverberation is generated in rooms whose walls are lined with sound-absorbing materials. In a room with a lot of reverberation, speech intelligibility is significantly impaired. For a successful music experience, however, the room should produce a certain amount of reverberation. The optimum decay time of reverberation depends on the type of music. Reverberation can be quantified using the reverberation time. The **reverberation time T_{60}** is the time in which the sound pressure level decreases by *60 dB* after the sound source has been silenced The evaluation of the reverberation time and its frequency dependence provides information on whether a room is suitable for speech and music performances.

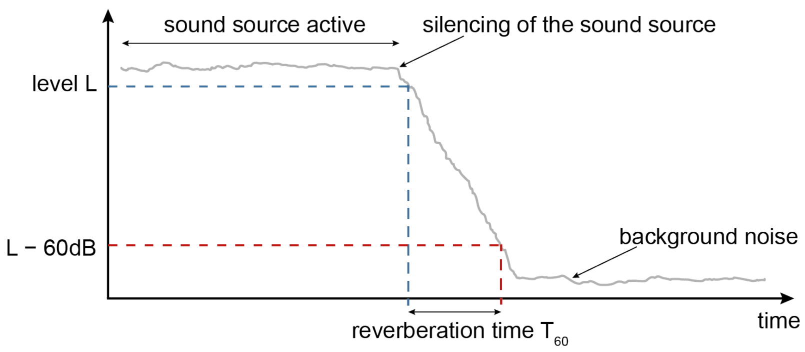


Figure 62: Determination of the reverberation

Reverberation chamber: A reverberation chamber is a room in which a diffused sound field can be implemented. It is constructed in a way that, ideally, the same sound pressure prevails at every point and thus the sound energy is evenly distributed in the room. In order to prevent the walls of a reverberation chamber from absorbing sound energy, they are made of sound-reflecting materials. This ensures that a large proportion of the sound waves are reflected by the wall. To avoid resonances, the walls and ceilings of the echo chamber are not aligned parallel. Furthermore, so-called diffusers are distributed throughout the room, by which the sound waves are reflected in all directions. The high diffusivity causes the sound waves in a reverberation chamber to be incident from all directions at the same time, so there is no directional sound pressure. Reverberation chambers are characterized by a strong reverberation and a long reverberation time. Reverberation chambers can be used to determine, for instance, the sound absorption of insulating materials and the sound power of sound sources.

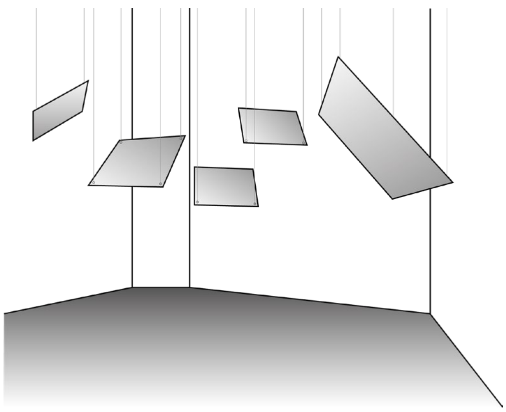


Figure 63: Reverberation chamber with diffussers

RJ45 connection: The RJ45 connection is used for data transmission in networks. The RJ45 plug has eight contacts

Figure 64: RJ45 plug

Root Mean Square, RMS: The RMS value is an averaged value of a time-varying quantity such as sound pressure. The RMS value depends on both the maximum value and the curve of the signal. The following formula applies to a sine tone with the maximum value *û*: $U_{RMS} = \frac{\hat{u}}{\sqrt{2}}$

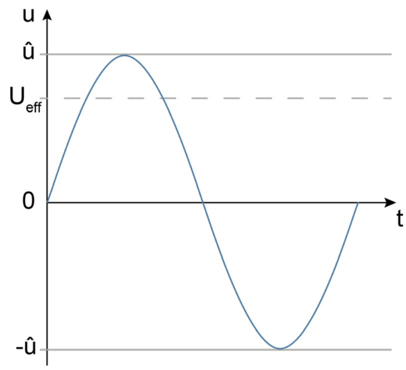


Figure 65: RMS value of a sine tone

Roughness: Roughness is a psychoacoustic parameter that is perceived in frequency and amplitude-modulated signals when the modulation rate is between *20* and *300 Hz*. The unit of roughness is *asper* (Latin: rough). A sine tone of *1 kHz* with a level of *60 dB,* which is amplitude-modulated at a modulation rate of *70 Hz* and a modulation factor of *1*, is attributed the roughness *1 asper*. With increasing roughness, noise emissions are perceived as being increasingly aggressive and annoying.

Sampling rate: For digital signal processing, time-continuous analog signals are converted into time-discrete signals. The sampling rate indicates the frequency with which an analog signal is sampled within a certain time.

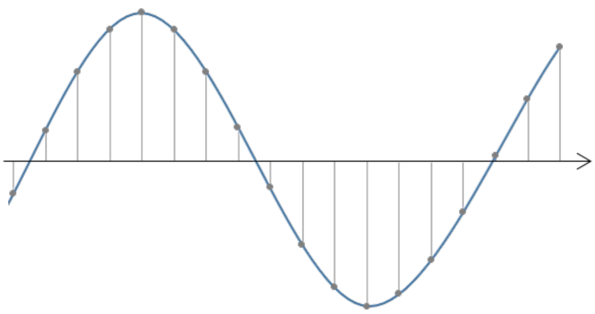


Figure 66: Analog time signal with equidistant sampling points

The sampling rate is given in the unit Hertz (*Hz*). The individual measurement results are called samples. At a sampling rate of *1 Hz*, one sample is measured per second. The higher the sampling rate, the more accurately the time curve of the original signal can be represented. A high sampling rate, however, also increases the amount of data to be processed. Sampling rates of *44.1 kHz*, *48 kHz* or *96 kHz* are often used for airborne sound signals. A lower sampling rate is usually selected for acceleration channels.

Sampling theorem: The sampling theorem is a fundamental theorem of signal processing. It states that a signal can only be clearly described by its sample data if the sampling rate f_s is at least twice as high as the highest frequency f_{max}: included in the signal: $f_s > 2 \cdot f_{max}$.
If a lower sampling rate is selected, this leads to the so-called aliasing. In contrast to the purely mathematical point of view, however, a sampling rate higher than twice as high has to be used in practice. This is due to the insufficiencies of real filter and quantization effects.

Sensitivity: The sensitivity is an important parameter of a sensor. It determines the transmission factor by which the measurement variable, e.g., sound pressure, is converted into an electrical voltage. The higher the sensitivity, the higher the output voltage for the same measured variable. The sensitivity of microphones is usually specified in the unit *millivolts per Pascal* $\left(\frac{mV}{Pa}\right)$. The sensitivity of accelerometers, for example, is specified in the unit $\frac{mV}{g}$ or $\frac{mV}{m/_{s^2}}$.

Sensor, Transducer: A sensor is a technical component that can detect a specific physical quantity and convert it into an electrical signal. Examples: a microphone to detect sound pressure; an accelerometer to detect acceleration.

Sharpness: Psychoacoustic sharpness is a perception quantity caused by high-frequency components in a sound. The unit of sharpness is *acum* (Latin: sharp). Sharpness is supposed to represent human perception linearly, i.e., doubling the acum value corresponds to doubling the sharpness perception. The value *1 acum* is assigned to a narrowband noise at *1 kHz* with a bandwidth of *160 Hz* (critical band width) and a level of *60 dB*. The calculation of sharpness is standardized in DIN 45692.

Signal-to-noise ratio, S/N: The signal-to-noise ratio is a measure of the technical quality of a useful signal that is superimposed by an unwanted noise signal. It is defined as the ratio of the average power of the useful signal to the average noise power. Good recording quality is ensured if the signal-to-noise ratio is sufficiently high and the useful signal is clearly distinguishable from the interfering noise.

Signal type: Time signals can be divided into different signal types. Various terms are used to characterize the signal type, some of which are explained below:

Stationary signals: Time signals are described as stationary if they only change insignificantly within the measurement period. For example, the sound of a constant drive on a level road is a stationary signal. For the analysis of stationary signals, analysis functions averaged over time can be used. (e.g., averaged FFT).

Transient signals: A transient time signal changes during the measurement. A vehicle interior noise recorded during rapid acceleration is a transient signal. Since transient signals change over time, a time-dependent analysis function should be used to analyze such signals (e.g., FFT versus time).

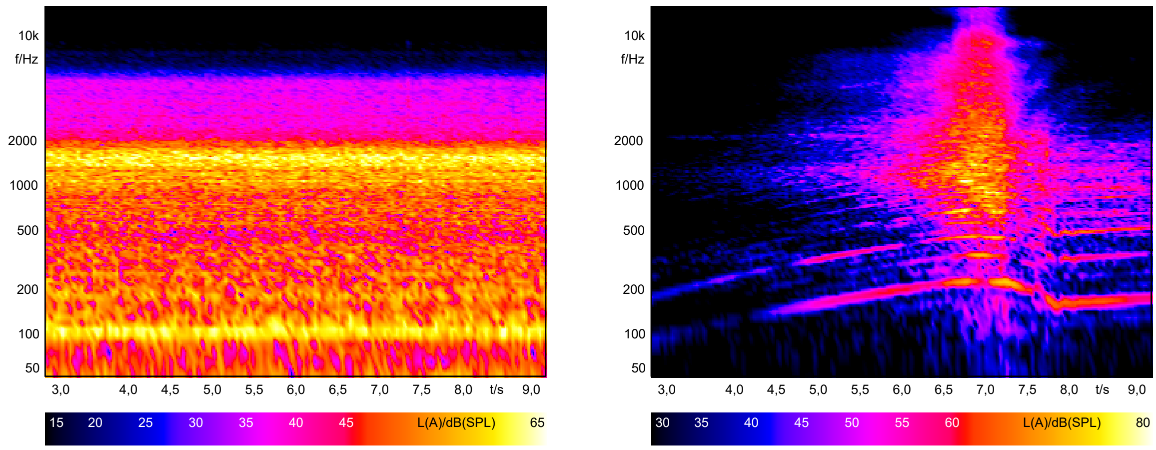


Figure 67: Stationary signal (constant drive; left) and transient signal (pass-by; right)

Stochastic signals: Stochastic signals follow a time course generated by random processes. White and pink noise are examples of technical, stochastic signals. Stochastic sounds are often caused by flows. Examples of stochastic sounds that occur in nature are water noise and wind noise.

Impulsive signals: Signals that change significantly within a short period of time are called impulsive signals. For example, a bang is an impulsive signal. An example of a periodic, impulsive signal is the knocking sound of a diesel combustion engine.

Sound absorption: Sound absorption reduces the sound energy, for example, by converting it into heat. Sound absorbing materials are used, among other things, to reduce sound reflections and thus reduce reverberation in rooms. The sound-absorbing properties of a material are quantitatively described by the frequency-dependent **sound absorption coefficient**.

Sound character: The sound character mirrors the auditory profile of a sound without considering product-specific information. Thus, in contrast to sound quality, the sound character describes the purely auditory perception, which exclusively includes the sensory aspects. Other factors, such as expectation or context, are not taken into account. This may cause a sound with a basically pleasant sound character to have a poor sound quality, if it is perceived as unsuitable for the product.

Sound design: Sound design refers to the analysis, design and processing of product sounds to optimize the sound quality. One goal of sound design can be, for example, to adapt the sound image to a given brand sound or to convey a certain product quality with the help of the product sound. In many cases, it makes sense to first modify the digitized sounds on the computer. These digitally performed modifications can be tested in listening tests, for example. In this way, desirable target sounds can be defined, which are then achieved, for example, through mechanical changes to the product.

Sound field: An area in which sound waves propagate is called a sound field.

Free field: In a free field, sound propagates undisturbed without reflections and diffraction. The perceived sound pressure is determined solely by the direct sound of a sound source. The free field is usually achieved in an anechoic chamber. Measurements in free field allow the properties of a sound source to be investigated without disturbing influences of the room, for example reflections, distorting the result.

Diffuse field: The diffuse field is a sound field with statistically equal probability of sound incidence from all directions. In an ideal diffuse field, the same sound pressure prevails at any location, except for the area directly around the sound source and directly in front of the walls. A diffuse field can be generated in reverberation chambers. A large proportion of the sound waves are reflected in these rooms, so that the sound energy is distributed evenly throughout the room.

The terms free field and diffuse field describe the acoustic properties of a room or environment. These properties are dependent on the respective room, not on the sound source.

Sound intensity: The sound intensity is the sound energy passing through an area element per unit time. It is calculated as the product of sound pressure and sound velocity. Consequently, the sound intensity is a vectorial quantity with both magnitude and direction. It indicates the energy flow in a sound field. The sound intensity of a spherical sound wave decreases by $1/r^2$ with increasing distance r from the sound source. Thus, if the distance doubles, the sound intensity is reduced to a quarter of the original value by quadrupling the surface of the sphere. The unit of sound intensity is *Watt per unit surface area* (*W/m²*). It is also common to indicate the sound intensity level in *dB*:

$L_I = 10 \cdot log_{10}\left(\frac{I}{I_0}\right)$ mit $I_0 = 10^{-12}\,\frac{W}{m^2}$

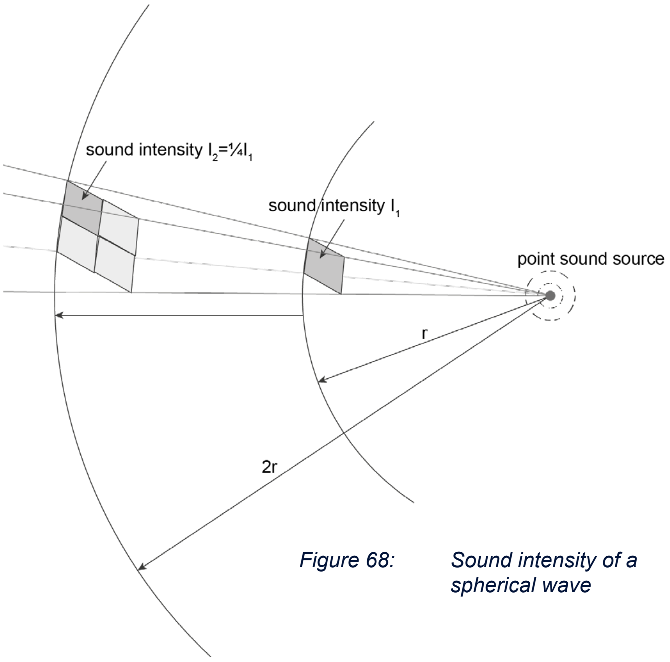


Figure 68: Sound intensity of a spherical wave

Sound intensity probe: Sound intensity probes are used to measure the sound intensity, though sound intensity can only be determined indirectly by simultaneously measuring sound pressure *p* and sound velocity *u*. Two types of probes are available to determine these two quantities: PU probes and PP probes. The PP probes, also called pressure gradient probes, measure the sound pressure *p* at two different points in the sound field by means of two microphones mounted close together. The pressure gradient can be determined from these two sound pressure values by linear approximation. From this pressure gradient the sound velocity is calculated by integration. The less common PU probes allow direct measurement of the sound velocity *u*. The velocity sensor consists of two heated wires lying close together. The temperature difference on these wires caused by the sound velocity combined with a change in electrical resistance provides a measurement signal proportional to the sound velocity. For measuring the sound pressure *p* the PU probe is also equipped with a built-in microphone.

Sound power: The sound power indicates the energy that a sound source emits per unit of time. The sound power thus describes the source strength of a sound source and not the sound field. In contrast to sound pressure and sound intensity, the sound power of a sound source is independent of the distance to the source. The unit of sound power is *Watt (W)*. It is also common to indicate the sound power level in dB: $L_P = 10 \cdot log_{10}\left(\frac{P}{P_0}\right)$ mit $P_0 = 10^{-12}W$

Sound pressure: The sound pressure $\tilde{p}$ describes the pressure fluctuation in a sound transmission medium. It is an alternating quantity that oscillates around the static pressure of the surrounding medium (e.g., ambient air pressure). The sound pressure is usually many orders of magnitude lower than the static air pressure.

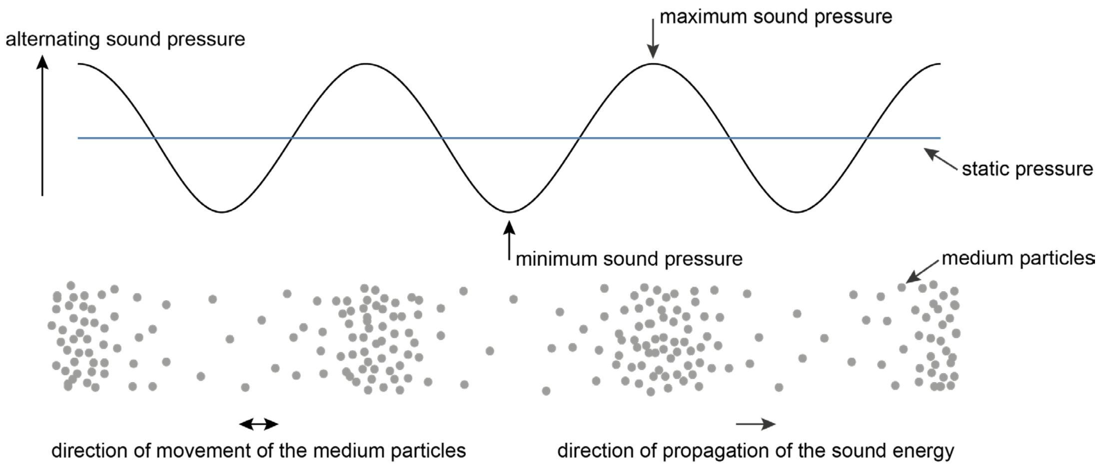


Figure 69: Generation of alternating sound pressure

The unit of sound pressure is Pascal with the unit symbol Pa. The sound pressure is a scalar, i.e., an undirected quantity. The sound pressure of a spherical sound wave decreases with increasing distance r from the sound source by 1/r. The sound pressure of a spherical sound wave therefore halves with a doubling of the distance.

Sound pressure level, SPL: The sound pressures, which can be perceived by the human ear, extend over a very wide range of values. Between the hearing threshold and the threshold of pain lie several powers of ten. In order to better handle this large range, the logarithmic representation is preferred. For this reason, the logarithmic sound pressure level *L* is determined from the sound pressure $\tilde{p}$: $L = 20 \cdot log_{10}\left(\frac{\tilde{p}}{p_0}\right)$ with $p_0 = 2 \cdot 10^{-5} Pa$

Sound pressure level is indicated in *decibels* with the unit symbol *dB*. The reference value is *0 dB* for a 1-kHz sinusoidal tone. This corresponds to a sound pressure of $2{\cdot}10^{-5}$ *Pa*. A doubling of the amplitude of the sound pressure leads to an increase in the sound pressure level of *6 dB*.

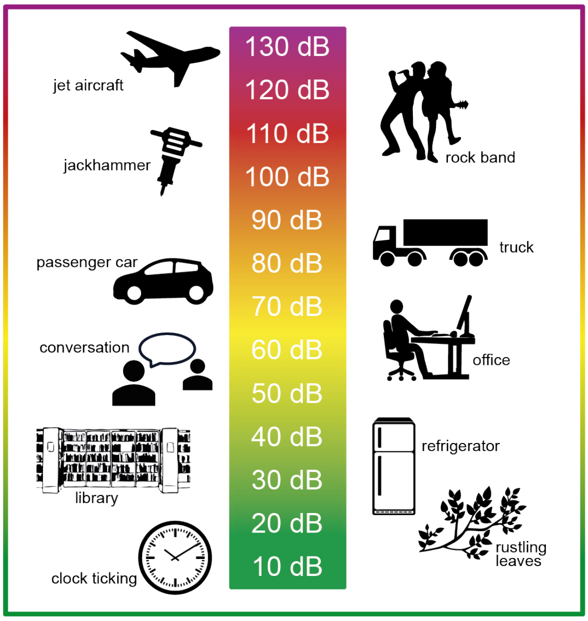


Figure 70: Decibel scale of sound pressure levels[1]

[1] The indicated sound pressure levels are only guide values. The actual sound pressure reaching the ear depends not only on the sound source, but also on the distance to the receiver!

Weighted sound pressure level: The human ear perceives sounds with the same sound pressure but different pitches at different loudnesses. This frequency-dependent sensitivity curve of the human ear can be simulated by weighting filters. In practice and in laws and regulations, e.g., for noise protection, the A-weighting curve is used almost exclusively. Although it was originally intended only for quiet noises, the A-weighting is now also used for louder noises. The C-weighting is also used for high sound pressure levels and for considering low-frequency noise components more intensively. A weighted level analysis, e.g., using the A-weighting filter, then shows the A-weighted sound pressure level.

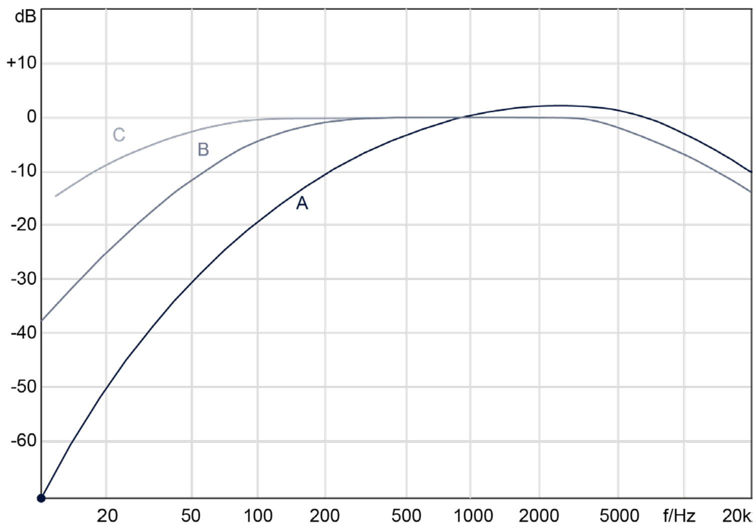
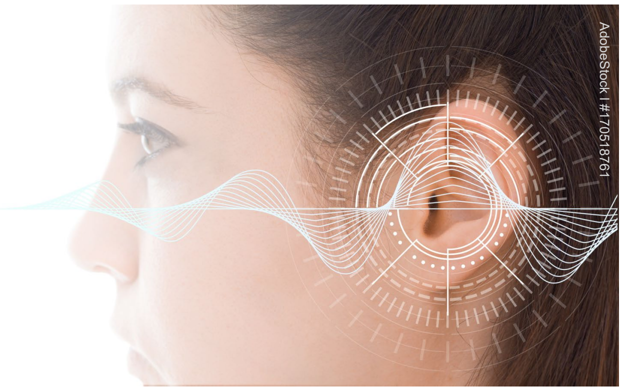


Figure 71: Weighting filters

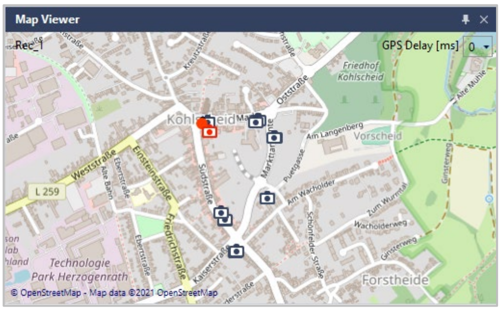
Sound quality: The sound quality describes the perceptive impression that is caused by a sound. A sound with good sound quality meets the user's expectations and is not perceived as disturbing. It creates positive associations with the product and thus improves the overall perception of product quality. A sound with poor sound quality, on the other hand, does not meet the user's expectations and is perceived as unpleasant, disturbing or annoying. This sound creates negative associations with the product and is perceived as unsuitable for the product. In most cases, the sound and its sound quality are used to assess product quality and functionality. A poor sound quality is often associated with poor product quality. In order to assess the sound quality, it is always necessary to also consider the context, since sound quality does not exist detached from context. The difference between good and poor sound quality is not, or not only, defined by the [sound pressure level].



Soundscape: Soundscape stands for an acoustic environment as it is perceived and experienced by a person. The perception of soundscapes and the reactions are highly context-dependent. Soundscapes are a complex phenomenon and therefore cannot be described by a single number. In order to record and analyze soundscapes, [artificial head] recordings are required, for example, as stated in the standard ISO 12913-2.

Sound source localization: When localizing a sound source, its direction and distance (both related to the listener) are determined. Humans can localize the sound sources in their environment by means of [binaural hearing].

Soundwalk: A soundwalk is an empirical method that is performed in-situ, i.e., during a walk in a specific acoustic environment. During the soundwalk the participants concentrate on listening. This is how context-sensitive data on [soundscape]s can be collected.



Sound wave: Sound propagates in a medium as wave resulting from pressure fluctuations caused by the oscillations of the medium particles. The oscillating particles of the sound transmission medium, e.g., air, do not move in bulk during sound propagation; they only perform an oscillating movement around their rest position. The wave merely carries the energy of the sound. The speed with which the energy is propagated is the **sound velocity**. In air at *20 °C* the sound velocity is *343 m/s*. The speed at which the medium particles oscillate around their rest position is called the **particle velocity** and does not correspond to the sound velocity.

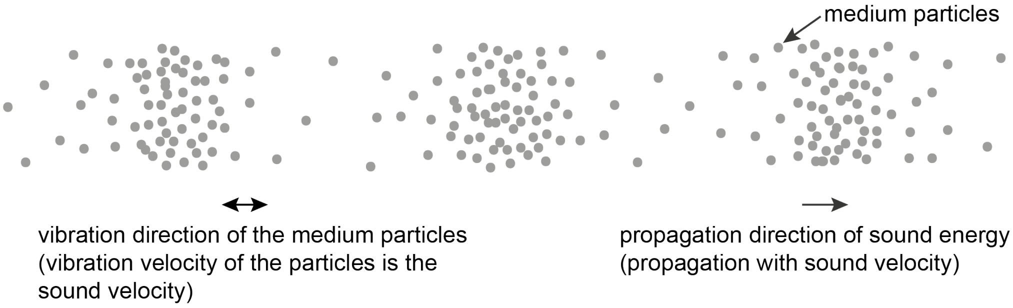


Figure 72: Propagation of a longitudinal sound wave in the medium air

In gases and liquids sound propagates as a longitudinal wave; the direction of oscillation of the medium particles is identical to the direction of propagation of the sound. In solids, transverse waves also occur in addition to longitudinal waves. With transverse waves, the particles oscillate transversely to the direction of propagation.
Sound waves can be reflected, diffracted and absorbed.

Plane wave: In a plane sound field, sound propagates in the form of plane waves only in one direction in space. Plane sound waves are characterized in that the propagating wave fronts are planes perpendicular to the direction of propagation.

Spherical wave: In a spherical sound field, the sound waves propagate in the form of a sphere. As the wave propagates, the energy of the spherical wave is distributed over an increasingly larger area. Therefore, the sound energy, related to the unit area, decreases with increasing distance from the sound source. In a spherical sound field, a distinction is made between the near field and the far field (near field: the distance to the sound source is significantly smaller than the wavelength; far field: the distance to the sound source is significantly larger than the wavelength). In near field, the vibrating surface of a sound source causes the adjacent part of the medium to vibrate, the medium is both moved as a whole (reactive power) and compressed and relaxed. In far field, the typical sound wave is formed in which the medium does not move in bulk and exhibits only compression and relaxation. The transition between the near and far fields is continuous. The further the spherical wave moves away from the sound source, the more similar the spherical waves become to plane waves.

S/PDIF: S/PDIF (Sony/Philips Digital Interface) is a specification for a unidirectional, self-synchronizing interface for the transmission of two or multi-channel digital audio signals. For the electrical transmission of signals in S/PDIF format, cinch connectors and coaxial cables with an impedance of *75 ohms* are usually used. It is also possible to transmit the data optically via TOSLINK.

Spectrum: see Frequency spectrum

Stationary wave (or standing wave): A stationary wave is a wave whose nodes of oscillation (zero crossings) and oscillation maxima always remain at the same location. It is created by the superposition of two waves of the same frequency and amplitude running in opposite directions; for example by a sound wave and its reflection running in opposite directions.

Structure-borne noise: Structure-borne noise is a vibration process in solid bodies. The sound is created by mechanical excitation of a body and passes through it as structure-borne noise. Pure structure-borne noise cannot be heard by humans but can be physically perceived. When structure-borne noise is converted into airborne noise by vibration of an object at a boundary with air, some of the structure-borne noise energy is emitted at the surface as airborne noise.

Timbre or tone color: The timbre of a sound is determined by the spectral and temporal energy distribution. For example, the timbre of two musical instruments may be significantly different even though they both have the same pitch. This is due to the different distribution of energy among the fundamental frequency and the harmonics as well as a different temporal structure of the sounds produced.

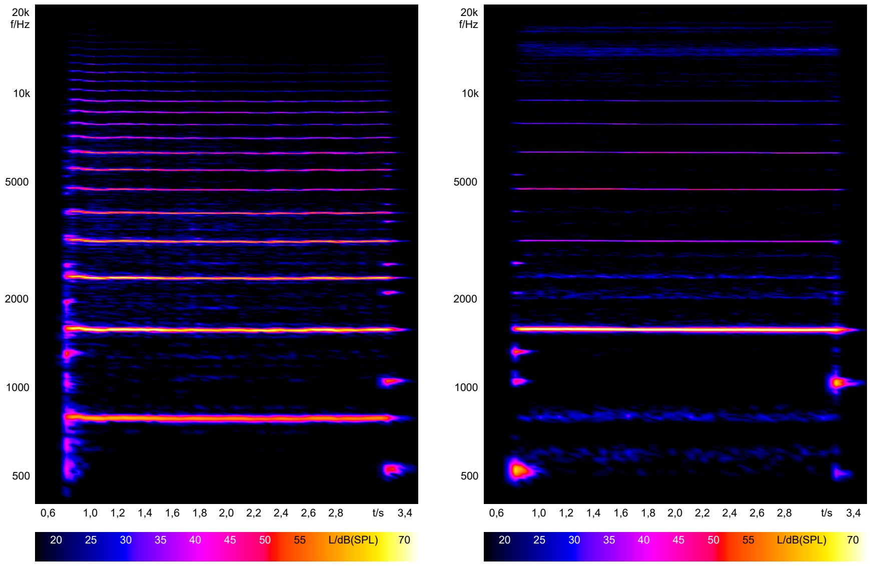


Figure 73: Spectrogram of the same note, played by two different music instruments (left: trumpet, right: flute)

Time weighting: Time weighting is an exponential smoothing of the sound pressure level with a certain time constant. The time constant used influences the subsequent analysis results. In principle, the larger the time constant, the stronger the smoothing. The influence is particularly clear with transient and impulsive signals.
Time weighting is used, among other things, to represent the sound pressure level over time.

Two constants are described in DIN EN 61672:2014-07:
fast: integration time: *125 ms*
slow: integration time *1000 ms*

However, the temporal resolution of the human auditory system is much finer. For this reason, a shorter integration time (approx. *2 ms*) should be used for a signal analysis that represents the human signal processing more accurately.

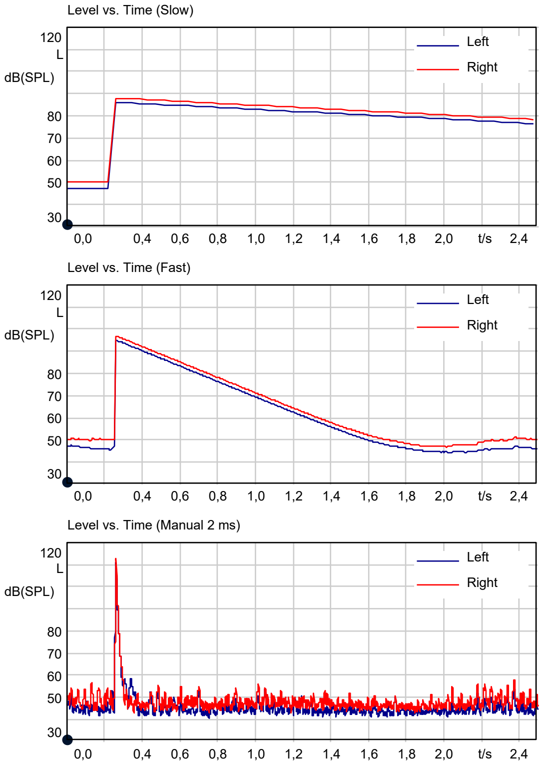


Figure 74: Level analysis of an impulsive signal with different integration times

Tonality: Sounds are perceived as tonal if they contain individual tones or narrow-band noise. Undesirable tonal sounds are perceived as more disturbing than comparable sounds without tonal components. In contrast, increasing the tonality of desired and pleasant sounds such as music can improve sound quality. A *1 kHz* sine tone with *40 dB* is given the value *1 tu$_{HMS}$* (tonality unit according to the Hearing Model of Sottek). The calculation method for the psychoacoustic tonality is described ,e.g., in the IT acoustics standard ECMA-418-2.

Torsional vibration: Torsional vibrations are rotational vibrations about an axis of rotation. Unwanted torsional vibrations are often caused by rotational irregularities and can lead to significant restrictions in the perceived quality of the product or even indicate a possible defect. In internal combustion engines, for example, torsional vibrations are caused by torsional irregularities of the crankshaft due to the stroke process of the pistons. But torsional vibrations can also occur in electric motors, for example, by torque ripple, i.e., a periodic increase and decrease of the torque output during rotation.

TOSLINK connection: The TOSLINK connector is a connection system for optical signal transmission with optical fibers. One advantage of optical signal transmission with optical fibers is that electrical isolation of the device components is achieved, thus avoiding ground loops. Optical fibers are also insensitive to electrical and magnetic interference.

Figure 75: TOSLINK connector

Transfer function: The transfer function describes the relationship between an input signal introduced into a linear, dynamic system and the output signal excited by it in the frequency domain. The transfer function required for modal analysis can be determined, for example, by simultaneously measuring the time signals of the exciting force and the response of the system to the applied force.

Transfer path analysis: Knowledge of the excitation signals is not the only important factor for noise optimization. It is also decisive how the different noise components contribute to the overall noise via the individual transmission paths. The aim of the transfer path analysis is to precisely investigate and describe the individual structure-borne and airborne sound transfer paths of a system, e.g., the transfer paths from the engine to the interior of a vehicle. In binaural transfer path analysis, the transfer functions of airborne and structure-borne sound are determined for the left and right ear of the listener. The structure-borne sound transfer path in particular is usually composed of several transfer paths. For a vehicle, the structure-borne sound transfer path from the engine to the vehicle interior can usually be made up of three transfer functions: Transmission of the engine mounts, apparent mass of the body, and acoustic transfer function to the driver's ear.

WAVE: The WAVE format is a file format for digital storage of audio data. In addition to the actual audio data, a WAVE file contains information about their format, such as the sampling rate. The file name extension of a WAVE file is .wav. Unlike MP3 files, WAVE files are usually uncompressed and therefore relatively large in size. The maximum file size of WAVE files is limited to 4 GB.

Wavelength: The wavelength λ of a sound wave is the smallest distance between two points of the same phase, i.e., two zero crossings passed through in the same direction.

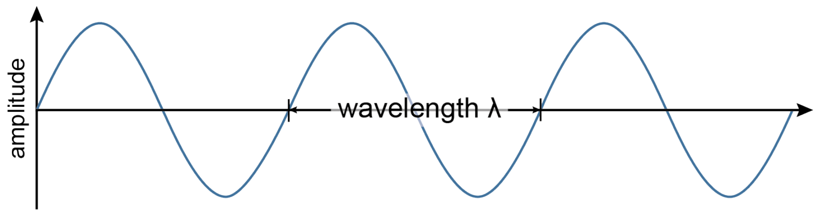


Figure 76: Wavelength

The wavelength is inversely proportional to the frequency of the wave, i.e., the higher the frequency, the smaller the wavelength. The following table shows the wavelength of sound waves of different frequencies (in dry air at *20 °C*).

Frequency	Wavelength
114 Hz	3 m
1,14 kHz	30 cm
11,43 kHz	3 cm

Weighted sound pressure level: see Sound pressure level

Window function: In digital signal processing, a window function weights individual signal sections prior to processing. Window functions are used, for example, in FFT analyses to reduce the leakage effect. For this purpose, each signal section is first multiplied by a window function that approaches zero towards the edges. This makes the end points of the signal section identical, resulting in a continuous signal without any points of discontinuity.
However, the use of a window function changes the resulting frequency spectrum and thus always represents a compromise. Since the selected window function affects the result of the analysis, it is important to select a window function that is suitable for the application at hand. For many applications the Hann window is suitable as it provides a good frequency resolution while simultaneously reducing the leakage effect very well. Some other window functions have been optimized for special applications. For example, the Flat-Top window provides high amplitude accuracy and is therefore suitable for calibration applications.
Other window functions were developed primarily to separate closely spaced frequencies, for example. To find a suitable window function for a particular application, the time signal can be examined in several runs with different window functions.

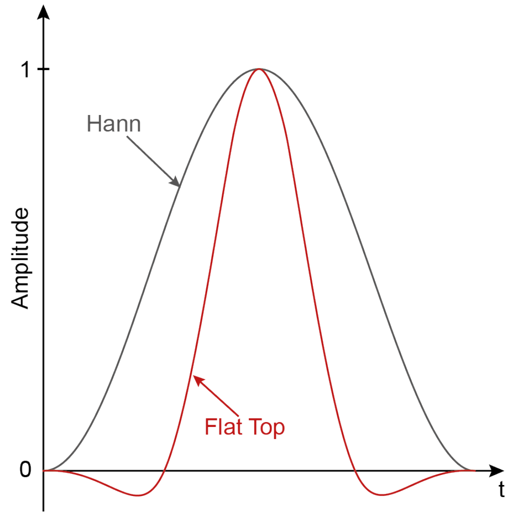


Figure 77: Hann and Flat-Top window displayed in the time domain

XLR connection: The XLR connection is an electrical connector for audio applications. It is used, for example, for professional analog audio signals, digital AES/EBU audio signals and also for power supply purposes. XLR connectors are available in three- to seven-pin versions. In contrast to cinch and jack plugs, the XLR connector has a locking mechanism.

Figure 78: Three-pin XLR connector