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TECHNICAL GLOSSARY

Acoustical Axis

The acoustical axis is a line normal to the loudspeaker's front panel along which the microphone was placed when tuning the loudspeaker's crossover during design. Pointing the acoustical axis, in the horizontal and vertical planes, towards the listening position or center of the listening area will give the best measured and perceived sound quality.

Acoustical Space (Free-field, Half-space, Quarter-space, and One-eighth-space)

A loudspeaker is said to be in "**free-field**" when it is placed in a space where there are no reflections or boundaries to restrict the sound radiation. These conditions can be found high in the sky (not so practical for listening or measurements) and in anechoic chambers. In practice, a loudspeaker placed on a stand far way from a room's walls can be considered to be in free space.

A loudspeaker is said to be in "**half-space**" when it is placed next to a large solid surface such as a wall or on the floor. The surface (acoustical boundary) limits the sound radiation to half of what it is in the free field. This has the effect of boosting the frequencies that are radiated omni-directionally, i.e. the low frequencies. The amount of boost depends on the solidity of the surface (theoretically 6 dB, in practice about 4 dB), and the frequency range depends on the size of the loudspeaker. This can be corrected with a suitably shaped filter, called "Bass" on Neumann loudspeakers. In practice, half-space is experienced when a loudspeaker is placed next to wall or flush mounted into a wall.

Similarly, a loudspeaker is said to be in "**quarter-space**" when it is placed next to two large solid surfaces such as a wall and a floor, or a front wall and a side wall. The surfaces limit the sound radiation to quarter of what it was in free-space. This has the effect of boosting the low frequencies twice as much as seen in half-space. This can be corrected with a suitably shaped filter ("Bass" and possible some "Mid" too), or in the case of subwoofers, attenuating the output.

"**One-eighth space**" is seen in the corner of rooms, and a very large boost is seen at low frequencies. This is a good location for subwoofers as the entire passband is boosted. This can be corrected by simply attenuating the output. For the same reason, loudspeakers generally sound very bassy in corners and so it is positioning that is not recommended for critical listening.

Active Crossovers

Active crossovers can be analog or digital. Analog crossovers are low-level electronic circuits positioned before the amplifiers in a loudspeaker system. As it is a low-level circuit, the filters can be precisely defined in terms of their frequency and phase responses. Digital crossovers are even more tightly defined and can offer additional signal processing that is either not practical or possible in analog electronics. Additionally, active crossovers do not heat up and change their properties as the loudspeaker is being used – this is a large source of distortion in passive systems. Klein + Hummel launched the world's first commercially available three-way loudspeaker with built-in active crossovers in 1967, the OY. These days the professional broadcast and recording industry has almost entirely converted to active technology.

Analog and Digital Signals

Analog is a continuous-time representation of a signal that is represented using an electrical voltage or an acoustical pressure. An analog input is required for analog signals. The electrical interface can be balanced or unbalanced. Balanced interfaces can be electronic or transformer based. There is a maximum input level beyond which clipping will occur. Various connectors are in use, for example RCA, Jack, and XLR.

Digital is a discrete-time representation of a signal that is expressed using numerical values. A digital input is required for digital signals. The input interface has parameters that must be appropriate for the signal: format, bit-depth, and sample rate. For example: AES3, 24-bit, 192 kHz. A practical example is, if the digital signal is always 48 kHz, as broadcasters always use, then a 48 kHz interface is sufficient. Various connectors are in use, for example RCA, BNC, and XLR.

Analog and DSP Loudspeakers

Analog loudspeakers have an analog crossover made using discrete components such as op-amps, resistors, and capacitors. The input to an analog loudspeaker can be analog or digital. Digital signals are immediately converted to analog using a digital-to-analog converter (DAC).

DSP loudspeakers have a crossover made using a digital signal processor and firmware. The input to a digital loudspeaker can be analog or digital. Analog signals are immediately converted to digital using an analog-to-digital converter (ADC). Digital signals can remain in their native sample rate and the internal filter coefficients changed to suit that sample rate. One coefficient set required for each sample rate so this is not very practical for all the sample rates now in use. These days a sample rate converter (SRC) is used to convert the digital input signal to a fixed internal sample rate. Additional hardware is required for this but the internal DSP implementation is much easier as only one set of coefficients is required. In the past, sample rate converters were not of sufficient quality to be used in high quality loudspeakers, but now they are.

Bass Management

Bass management moves the low-frequency portion of a signal to a loudspeaker other than the one normally used to reproduce that channel. For example the bass of the center channel is reproduced by a subwoofer. This reduces the work that the center loudspeaker is required to do thereby allowing a smaller loudspeaker to be used or for the same sized loudspeaker to play with reduced distortion or at a higher level. Additionally, the LFE-channel is routed to subwoofers and/or loudspeakers that have the capacity to reproduce this high-level low-frequency signal. Finally, greater flexibility is available when positioning the source of the low-frequency energy in the room which can result in a better sound quality.

Decibels (dB)

The human auditory system is a logarithmic device, so the numbers used to describe audio perception can easily become quite small or large and thus tricky to handle. Converting these logarithmic-spaced numbers to a linear scale is more convenient to work with and read. Decibels have no dimensions as it is the ratio of two values having the same dimensions. The classical Bel calculation gives values that are quite small for practical purposes, so a factor of 10 is used to give decibels.

$$W = 10 \log_{10} (P/P_{ref}) \text{ dBW}$$
$$\text{where } P_{ref} = 1 \text{ Watt}$$

Some simplified examples: An amplifier of 100 W can be expressed as 20 dBW. If a loudspeaker driver has a sensitivity of 90 dB/W/m, the maximum output with a 100 W amplifier would be 90 + 20 = 110 dB SPL. If the listening position is at a distance of 4 m, the maximum SPL would be 110 – 12 = 98 dB SPL as sound pressure halves with each doubling of distance.

An additional factor of 2 is used when the ratio values are pressure or voltage, rather than energy or power.

$$SPL= 20 \log_{10} (p/p_{ref}) \text{ dB SPL}$$
$$\text{where } p_{ref} = 2\text{x}10^{-5} \text{ Pascals}$$

Some simplified examples: A sound pressure of $2\text{x}10^{-5}$ Pa, which is the threshold of hearing and used as a reference level, is easier to express as 0 dB SPL. Then 1 Pa is 94 dB SPL and 130 dB, which is the threshold of pain, is 64 Pa. If two loudspeakers are playing at 94 dB SPL, the total level would be 100 dB SPL.

Frequency Response Limits

There are a number of ways to describe the low and high cut-off frequencies of a loudspeaker:

The **−3 dB** points are defined as the frequencies where the frequency response is 3 dB lower than the average level of the pass band of the loudspeaker. These are the most often quoted values and are easily comparable from one loudspeaker to another, even models from different manufacturers. An example is 30 – 24k Hz ±3 dB.

The **passband** points are defined as the frequencies where the low frequency response passes through the lower pass band tolerance level. This value defines the lowest frequency where the loudspeaker reproduces audio with the same accuracy as the rest of the response. It is not useful for comparing different loudspeakers unless they have the same passband tolerance. If the passband tolerance is less than ±3 dB, the specified frequencies will be inside the ±3 dB point frequencies. An example for the same loudspeaker specified above is 32 – 20k Hz ±2 dB.

The **useable operating frequency range** is defined by the two frequencies that are 10 dB lower than the average level of the pass band of the loudspeaker. Whilst this may seem to be a very wide tolerance, it is a frequency range of an unequalized loudspeaker that can be corrected using equalization. This specification is commonly seen in loudspeakers for use in installations and live sound. An example is 58 – 22k Hz for a loudspeaker that has a ±3 dB frequency response of 90 – 19k Hz.

Level Weighting (Lin, A, B, C, D, Z)

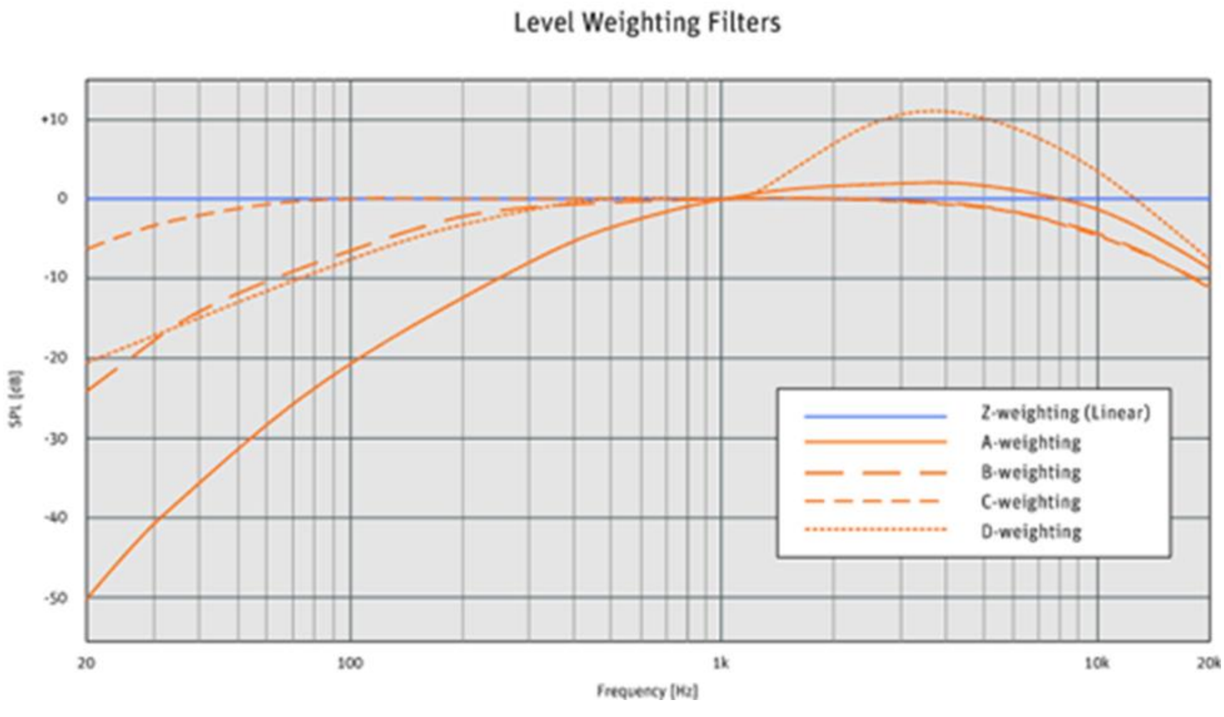
A measurement of sound pressure level can either be performed at particular frequencies or as a wide band summation over a defined frequency range, say 100 – 6k Hz. The ear is not equally sensitive at all frequencies and so an emulation of this is required to get an idea of the level of a sound. Additionally, the shape of the ear's sensitivity changes depending on the level of the sound: it becomes flatter at higher levels. The frequency shapes (weightings) are defined in IEC 61672:2003. The different weightings were designed to be used for different sound levels but laziness has meant that only the A and C -weightings are typically used, thus creating a ubiquity.

"Linear" or "**Lin**" is used for peak SPL measurements. Sometimes LP and/or HP filters are inserted.
"**A**" is used for measuring sounds with a low SPL (40 phon). The low frequencies are heavily attenuated and the high frequencies are moderately attenuated.
"**B**" is used for measuring sounds with a medium SPL (70 phon). It is not used very often.
"**C**" is used for measuring sounds with a high SPL (100 phon). It has a gently rolling off low and high frequency response.
"**D**" and "E" are used for measuring sounds with a very high SPL, like aircraft noise.
"**Z**" or "**Zero**" is used for peak SPL measurements. LP and/or HP filters are not allowed.

Other weightings exist for special purposes like measuring infrasound.

Additionally, time-domain averaging is used: slow (1 s), fast (1/8 s), and integrate. "Slow" is typically used for tuning sound systems as the display changes relatively slowly thereby allowing time to take a reading of a rapidly time-varying signal such as music. "Fast" and "Integrate" are used by consultants for noise assessments.

To measure a sound system, "C-weighting" and "Slow" is generally used.



Low Resonance Integral Molding™

The entire front panel is constructed using Low Resonance Integral Molding™ materials (LRIM™). The midrange and treble drivers are mounted into a waveguide – see Mathematically Modeled Dispersion™. The waveguide is injection molded. The rest of the front panel is either a part of the waveguide molding or CNC machined.

Magnetic Shielding

A loudspeaker driver's motor consists of a coil moving inside a permanent magnet. The magnet has a usable magnetic field where it is required (i.e. where the voice is located), and a stray magnetic field where is not required (i.e. around the outside of the driver). Unless the loudspeaker cabinet made of steel, the stray magnetic field will be experienced outside the cabinet. This will distort the picture on CRT screens and potentially destroy data on magnetic storage media, such as hard drives and tapes. To cancel most of the stray magnetic field a bucking magnet (a smaller magnet with the opposite polarity) is glued onto the back of the main magnet.

Sealed and Vented Cabinets

A vented cabinet has a tube somewhere in the cabinet. The acoustical system is then resonant at certain frequencies. This has the effect of producing a lot of sound for not much movement of the bass driver's cone. The advantages are a reduced bass distortion, a higher maximum SPL, and a smaller required amplifier power. The disadvantages are a higher group delay at low frequencies, complex port design required, potential port turbulence and noise, and space is required for the port and its opening.

A sealed cabinet design has no ports or vents, so there is a limit to the output level at low frequency. The disadvantages are an increased bass distortion, a lower maximum SPL, and a larger required amplifier power. The advantages are a lower group delay at low frequencies, no port design required, no port turbulence and noise, and no space required for the port and its opening.

Except for the openings of vent in a vented design, both cabinet types must be completely air tight. Small openings can create whistling sounds and compromise the acoustical performance of the system.