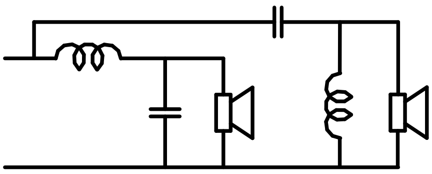
11.11 Studio-monitors

In the control room of a recording studio, high-grade multipath loudspeakers are deployed. Especially during the 1950's to 1970's, they were often fitted with mid- and treble-range horns. Contrary to widespread opinion, the frequency response measured on-axis is not the most important criterion. The frequency dependency of the "free-field transfer function" is not unimportant, but premium loudspeakers handle this aspect so well that other criteria move to the focus, for example the beaming, or (at high monitoring volume) the distortion (THD, difference tones, sub-harmonics). Because satisfactorily handling the whole audible frequency range with a single loudspeaker is not possible, filters (crossovers) take care of a subdivision into several frequency bands fed to corresponding speakers. Shown in **Fig. 11.118** is a simple circuit, as it is found (with slight modifications) in many DIY-guides. For the corresponding calculation it is assumed that the loudspeaker impedance is real, and that impedance and transmission-factors are frequency-independent. These assumptions are far from reality: the impedance is complex and dependent on frequency (Fig. 11.9), as are the transmission factors (in particular the phase). However, let us follow for a moment the idealized train of thought: the 2nd-order low-pass shifts the phase from 0° to -180°, and the 2nd-order high-pass generates a shift from 180° to 0° – such that across the whole frequency range the speaker voltages are in opposite phase. Only connecting the speakers 'out-of-phase' will avoid a complete cancellation at the crossover frequency (600 Hz in our example). That an all-pass filter results is, on the other hand, not critical: out hearing system does not take notice of that [3].



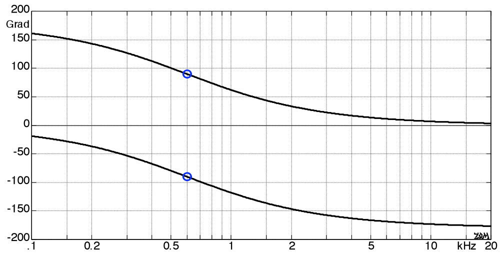
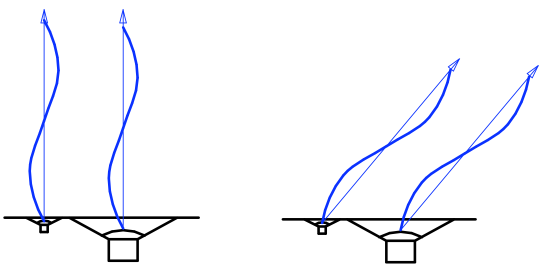


Fig. 11.118: 2nd-order two-way crossover: circuit (left), frequency response of the phase (right).

The big problem is created at the crossover frequency, if both speakers radiate the sound with the same amplitude. Even if the two partial sounds sum up perfectly on-axis – the radiation towards the sides always involves a phase shift, creating a destructive interference. If the difference in the path-length corresponds to half the wavelength ($\lambda = c/f$), the partial sounds cancel each other out (**Fig. 11.119**). An improvement is possible via so-called coaxial systems with the woofer being positioned behind the mid-range-speaker on the same axis; however here the speakers may get in each other's way. The argument that we should simply listen only exactly in front of the speaker does not hold water, either: the reflections arriving from the side do influence the hearing perception, as well.



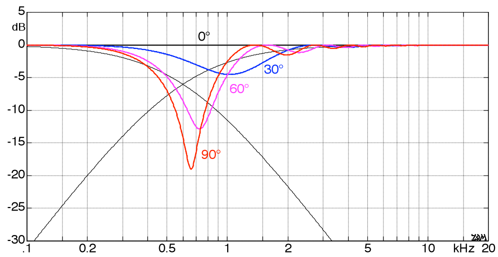


Fig. 11.119: Interference in the crossover-frequency-range: cancellation for radiation towards the side.

The following examples show the beaming behavior of different 3-way-speakers. We see from **Fig. 11.120** how even well known manufacturers struggle: the often requested “monotonous increase” of the directivity is nowhere in sight.

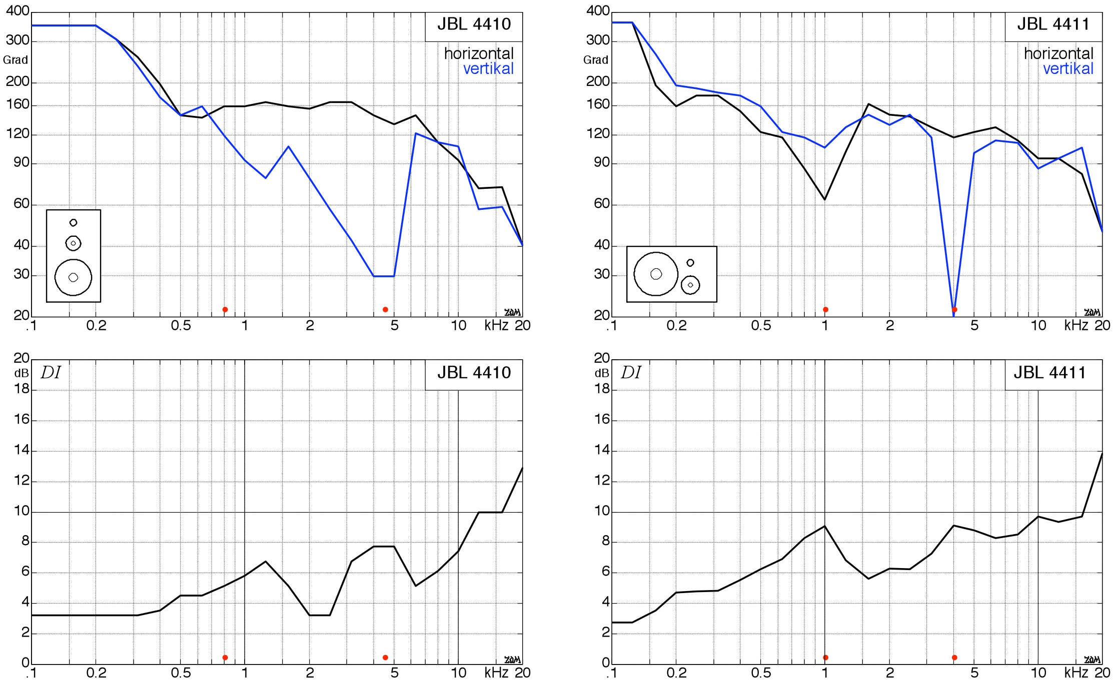


Fig. 11.120: Aperture angle and directivity of two 3-way-speakers (according to the manufacturer’s datasheet). “Vertikal” = vertical

The data of Sentry III are shown in **Fig. 11.121**; this speaker already enjoys a cult-status, and not undeservedly, as the graphs indicate. Still, we need to note that the two frequency responses of the aperture angle are always only simplified representations of a highly complex beaming behavior (Fig. 11.111). Also, hearsay states that there may be manufacturers who will “lend some help” to a less optimal curve and conjure up a characteristic favored by the sales department.

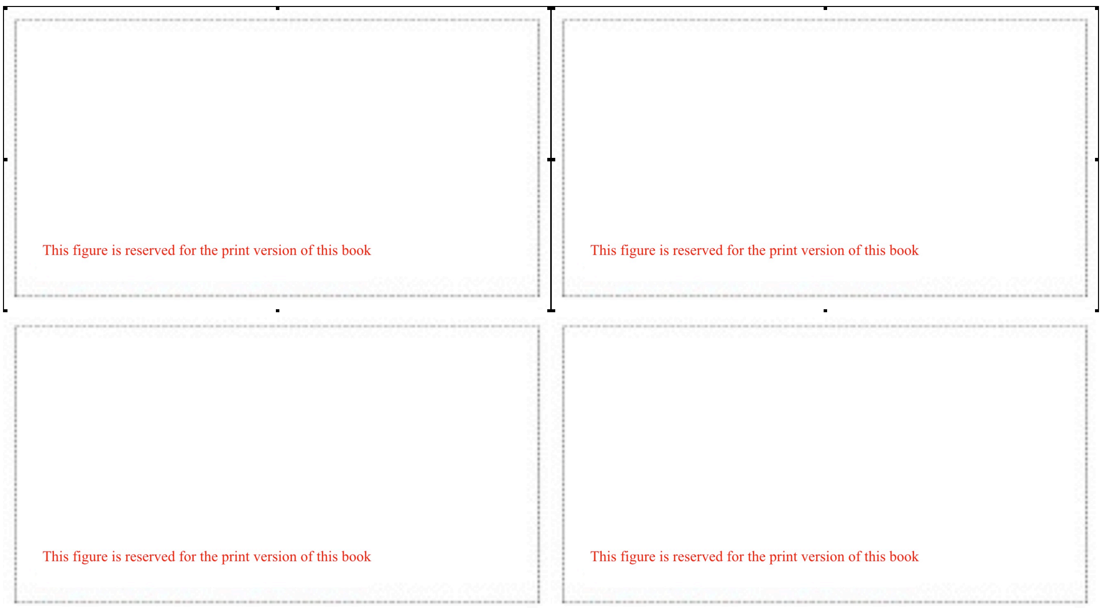


Fig. 11.121: Aperture angle and directivity of two 3-way-speakers (according to the manufacturer’s datasheet).

The following figures (**Fig. 11.122**) belong to two-way speakers. The JBL and the Altec can easily be imagined placed in the studio while the EV-speaker is more intended for PA-use. The 604-8L combines a 15"-woofer with a Mantaray-horn mounted coaxially with the woofer; the two JBL's employ so-called bi-radial horns (100°x100°), and the EV-box sports a 90°x40°-horn. None of the directional characteristics could be designated as particularly good or particularly bad – the quality always depends on the individual deployment-location. In the studio, this will be a relatively strongly absorbent control room where the reverberation time is between 0.2 and 0.4 s resulting in a reverberation radius of about 1.5 m. The effective reverberation radius [3] will then be around 2 – 6 m, and that will give the diffuse sound some significance, after all.

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Fig. 11.122: Aperture angle and directivity of two 3-way-speakers (according to the manufacturer's datasheet).

Fig. 11.123 shows the reverberation time $T_{60}(f)$ of two professional control rooms. For one of them (black curve), the right-hand graph indicates the effective reverberation radii, i.e. the reverberation radii increased by the square root of the directivity factor [3]. An engineer listening back at a distance of 3 – 4 m from the speakers is therefore predominantly located in the diffuse field for low frequencies. Depending on the beaming of the speakers, a very special mixture of direct and diffuse sound results that turns out to be rather … shall we say: "characteristic" for the JBL 4425.

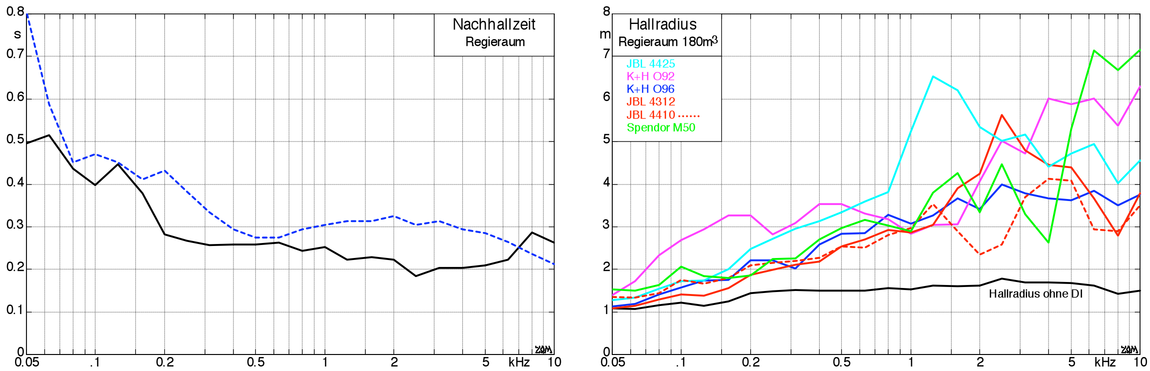

Fig. 11.123: Frequency responses of the reverberation times in two control rooms (left).
The black curve in the right-hand graph indicates the effective reverberation radii for 6 different studio monitors.

As a conclusion, let us look at a few measurements regarding **non-linear distortion** (**Fig. 11.124**). The requirement to be able to generate an SPL of 80 dB at distance of 2 m is not a very challenging one. However, if the maximum harmonic distortion needs to be kept below 0.1%, a few speakers fail, after all. Your classical studio-monitor will be able to rather reach around 1% – that is not all that bad, but more modern, newly developed types are able to remain clearly below the 1%-mark. Of course, the 80-dB-@-2-m is not the maximum required SPL – that would be about 110 dB / 2m. But even at that level, the non-linear distortion should remain "inconspicuous".

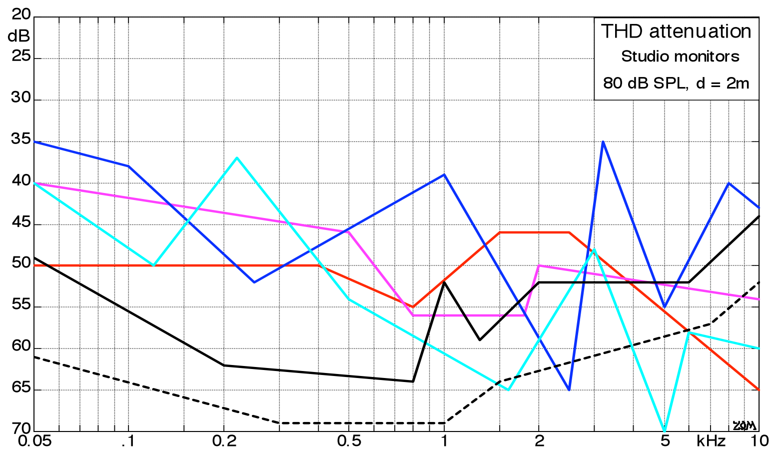

Fig. 11.124: Harmonic distortion suppression of different studio monitors (according to the manufacturer's datasheet).