Studio monitor midrange and high frequency performance

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With the aim of improving the sound quality of the loudspeaker system, it is essential to have a working understanding of the problems associated with sound reproduction. The most important parameter in describing the perceived sound quality is the frequency response. The smoother the spatially averaged frequency response in the listening area, the higher the fidelity rating. A flat on-axis frequency response is all very good, but of little use to a recording engineer moving along the mixing desk.

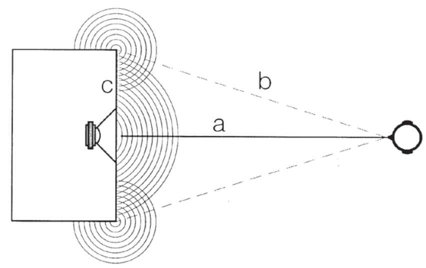

Figure 1a. Sources of secondary radiation for on-axis listening. Path difference between direct and secondary source = b + c - a.

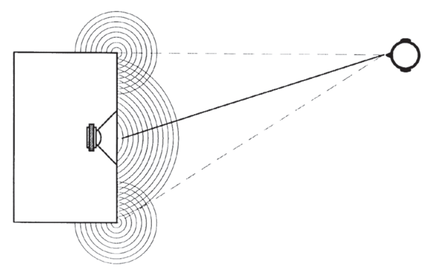

Figure 1b. Sources of secondary radiation for off-axis listening.

Diffraction

One of the important phenomena affecting sound radiation of a loudspeaker is diffraction. Explained simply this means that any edges, for example the loudspeaker cabinet edges, act as secondary sound sources. As the cabinet front panel usually has four edges, the system has five radiators operating at the same time, the actual driver and four secondary sources. The resulting frequency response at the listening position is the sum of all these sources (Figure 1a). The summed frequency response varies because of the summing of components with different arrival times. As the listener moves off-axis (Figure 1b), the relative distances of the secondary sources change. The summed response heard at the listening position is different due to the secondary radiations. The existence of secondary sources degrades the response of a loudspeaker. On-axis there is ripple due to the summing of coincident edge diffractions. Off-axis the summing is no longer coincident as the path lengths are different, which results in lower amplitude of ripple spread more evenly over frequency. This is usually seen at and above midrange frequencies due to the physical dimensions of loudspeaker compared to the wavelength of the radiated sound.

A reduction in sound level occurs when the difference of the direct radiation path and the path length of the diffracted sound equals half the wavelength (λ)

$$b + c - a = \lambda/2.$$

Loudspeaker cabinet diffraction has been carefully studied in acoustical textbooks (Muller et al. (1938) and Olson (1947)). The box shape and the driver location are key factors.

Effects of diffraction

To reduce the frequency response ripple the distances from the driver to cabinet edges can be made unequal so that the ripple pattern is distributed more evenly over frequency. Off-axis the frequency responses of left and right speakers will then differ blurring the stereo image. To avoid this many speakers of this type are built in mirror pairs.

A very useful and simple test of stereo performance is to use a mono signal. The centre image should be sharp at all frequencies. Reproduction of a mono signal is a simple way to check the system’s ability to reproduce phantom images in general. If the mono image is wide and frequency dependent, then the stereo performance is usually also bad. Diffraction is not the only reason for poor mono performance, but other reasons are beyond the scope of this article.

Listening environment

The loudspeaker itself is not the only thing in the control room that can cause frequency response problems. In fact, there are numerous objects and surfaces that reflect sound energy towards the listening site. It does not matter what the cause of a delayed signal is. Sound quality is degraded if there is a delayed signal which arrives shortly after the main sound from a direction close to the direct sound (Haas effect).

Ideally the control room environment should be clear of any equipment that might cause interfering reflections. The mixing console and other rack equipment cause reflections. Traditional two and three way designs work in geometrically well designed control rooms which are sufficiently damped. In poorer environments their responses may be degraded due to room reflections.

Power response

The power response is the loudspeakers total radiated output, not just on-axis, but in all direction around it. It can be measured in a number of ways (Olson, 1947), for example by measuring sound pressure level in an anechoic chamber at various angles along a sphere (or hemisphere), and integrating the results.

Listening in a control room the engineer hears both the direct sound and the reverberant field. The perceived frequency balance at the listening position is linked to the power response of the system. The more omnidirectional the loudspeaker, the more the listener will hear of the room's reverberant field.

Let us now imagine a true omnidirectional loudspeaker with flat on-axis frequency response. It radiates identically

Haas effect

Precedence effect, also called Haas effect after its inventor, refers to the ability of man to hear a sound that is emitted shortly after the main sound, for example a wall reflection or a diffraction from the loudspeaker cabinet edge.

We do not hear a separate sound if the **delay** of the second sound is less than about 30 ms, corresponding to a time-of-flight for the sound of 10 meters (Buser, 1992). If the delay is smaller, then the original sound takes **precedence** over the secondary sound. This is called the Haas effect.

The Haas effect several important consequencies for loudspeaker design. If the delay of the wall, floor and ceiling reflections is smaller than 30 ms, these can not be discerned as separate sound. Instead they will **modify** the frequency balance of the original sound. These reflections have a special name, they are called the **early reflections** because they integrate with the main sound modifying the sound quality. Because of this, even a perfect loudspeaker without the DCW will be spoiled in a room with significant early reflections.

Balancing the intensity with time

When we listen to a loudspeaker stereo pair, we hear a sound event in the middle between the two speakers, if the distance to both speakers is equal and the level of the sound input to both speakers is the same. If we change either the input level or the time delay to one of the speakers, the sound event appears to move to a new location.

Sound travels at a speed of 340 meters/second distance of one foot (about 30 cm) corresponds to about 1ms of time-of-travel. If you move one foot closer to one of your speakers, keeping the distance to the other constant, you modify the delay difference between the two speakers by one millisecond. The ear is sensitive to this delay difference, and you experience the location of your sound event to move in front of you. The input level to your speakers also moves the sound event. This is very familiar to us. We use the balance potentiometer to move the sound in our stereo system where we want to place it.

To an extent it is possible to compensate the delay change by a change in level. Many investigators have measured the value of this time-intensity trade-off, and it is typically expressed in microseconds/dB. This trade-off is frequency dependent, and the typical values are between 25…90 microseconds/dB (Moore, 1989). The DCW design and the Genelec recommended way to aim the monitors constructively exploits the time-to-intensity trade in the human hearing system to provide a stable and wide listening area.

in all directions and has a flat power response. In reality loudspeakers are only omnidirectional when the driver's physical dimensions are small compared to the wavelength of the radiated signal. As frequency increases (wavelength shortens) the radiator becomes increasingly directive. Consider a typical system consisting of a 250mm woofer and a 25mm tweeter. The woofer becomes increasingly directive with frequency and its radiation angle becomes narrow at the typical crossover frequency of about 2kHz. At this point there will be a radical change because the tweeter is virtually nondirective at this frequency. The power response will then peak up at the crossover point. As frequency increases the tweeter becomes more directive and the power response begins to decrease. If no effort is made to correct this and the on-axis response remains flat, the net result will be an uneven power response. The practical meaning of this is that when a peak occurs the proportion of the direct sound to the total sound output decreases. The net effect depends on the listening environment, resulting in the perceived frequency balance varying from room to room.

Figure 3. The Genelec 1038A DCW.

Optimizing in-room performance

To avoid problems associated with diffraction the driver maybe flush mounted into a baffle. We can use this principle by flush mounting the whole loudspeaker into the control room wall. Flush or soffit mounting also has the added advantage of improving the low frequency response through avoiding a back wall reflection.

An alternative solution to soffit mounting could be considered. If there is no radiation towards the diffracting edge, nothing can reradiate from it. This can also be achieved by making the cabinet edges round at the frequencies of interest.

At mid and high frequencies the radiation angle can also be limited by a waveguide structure to avoid diffraction from cabinet edges. The benefits of a limited and controlled radiation angle become obvious as the acoustical conditions worsen. The controlled directivity improves the ratio of direct sound to early room reflections. The listener is more in the direct field and is able to hear more of the program material and less room effects. Subjectively this is perceived as improved imaging and better definition. If the radiation pattern is constant power response will be uniform and flat. The system will also be more immune to changes in room environment.

Directivity Control Waveguide

Genelec has developed the Directivity Control Waveguide technology already in 1982, and it is used in all our new full range monitoring designs. DCW is a novel acoustical device, which shapes the emitted wavefront allowing control of dispersion. It is a specially curved rigid surface fitted in front of the driver unit. It can be dimensioned for constant directivity extending down to fairly low frequencies depending on the DCW frontal dimensions. The DCW allows cabinets with sharp edges to be used without problematic diffraction effects.

Wide Stereo listening area

The main monitors are traditionally aimed at a focal point behind the engineer's position, with equal distances between the speakers and the listener. The human ear uses both amplitude and time cues in the localisation of sound sources. The stereo imaging depends on both time and amplitude differences between the left and right signals. Moving to either side of the centreline between the speakers causes an image shift to the nearest loudspeaker.

Imagine a system where the dispersion pattern is constant and controlled at all frequencies. Its off-axis response is flat, but the level is lower than on-axis. We aim this pair at or in front of the listening position. Now assume that the listener moves to the right of the centreline. As he moves off-axis, the signal level decreases. At the same time the distance to the right loudspeaker shortens and the level slightly increases. The listener moves increasingly on-axis of the left-hand loudspeaker. The level of the left loudspeaker increases. The net result is that it is possible to aim the speakers in such a way that the imaging remains more stable although the listener moves slightly off the centreline. In this way the DCW systems create a wide stereo listening area. Also the power response becomes more uniform without peaks or dips at the crossover (Figure 4). The DCW can be made an integral part of the enclosure construction.

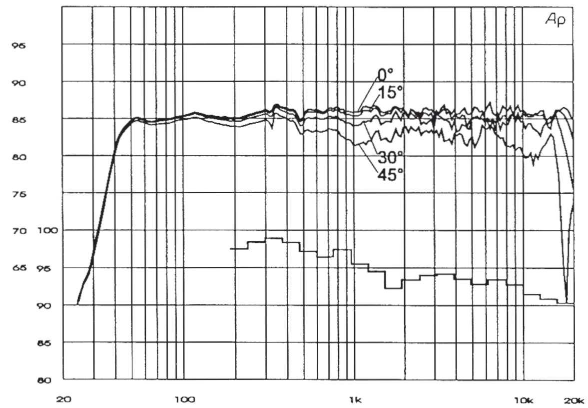

Figure 4. Typical free standing DCW monitor frequency responses. The lower curve is the power response of the system, measured in an IEC reverberant chamber.

High sensitivity and low distortion

Limiting the radiation space angle boosts the output. The sensitivity of a good DCW design can be up to 10dB better than the sensitivity of a typical direct radiating driver.

The distortion of a DCW loaded driver can be very low. For example, Genelec has achieved harmonic distortion of less than 0.5% between 500Hz and 4kHz at 110dB SPL (Table 1). This is one tenth of the distortion of compression drivers at the same SPL.

f (Hz)	d_2 (%)	d_3 (%)
500	0.2	<0.1
1000	0.25	<0.1
2000	0.3	<0.1
4000	0.4	<0.1

Table 1. Harmonic distortion figure of the Genelec midrange driver at 110dB continuous SPL measured at 1m.

The DCW technology allows to control the directivity pattern in a predictable way. The frequency response becomes uniform in the listening area. Diffraction and distortion are minimised as the sensitivity increases. All of these aspects greatly enhance the overall fidelity of the system allowing reliable results in vastly differing acoustic environments.

References

Buser, Imbert: Audition. MIT Press, 1992. 394p.

Hassel, Zaveri: Acoustic Noise Measurements. Brüel & Kjær, 4th edition, 1079

Martikainen, Varla, Partanen: Design of a high power active control room monitor. Audio Eng. Soc., 86th convention, March 1989, Pre-print 2755 (J-2)

Moore: An introduction to the psychology of hearing. Academic Press, 1989. 350p.

Muller, Black and Dunn, J. Acous. Soc. Amer., Vol.10, No. 10, p.6, 1938

Olson: Elements of Acoustics. D. Van Nostrand Inc., 1947, p.20-23

Toole: Loudspeaker measurements and their relationship to listener preferences: Part 2. J. Audio Eng. Soc., Vol. 34, No.5, pp. 323-348, May 1986