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Audio Reproduction: Why Simplicity is Best

The design of loudspeakers is a constantly evolving technology with multiple parallel forks that are all working toward a single goal – identical acoustic reproduction. They are one of the most omnipresent objects in our lives yet professionals agree wholeheartedly on their vast array of imperfections. Loudspeakers swarm our public spaces, private spaces, entertainment systems, mobile devices, automobiles, and outdoor gatherings, but all fall well short of true accuracy. Complex issues such as off-axis frequency falloff, cone breakup, doppler distortion, and phase linearity, all wreak acoustic havoc atop seemingly simpler criteria such as frequency response, crossover design, baffle step diffraction, and physical placement. For simplicity of scope we will limit ourselves to a simple comparison of factors between a single full-range hi-fi loudspeaker and a comparable multi-way hi-fi loudspeaker. Theoretically, both are intended for domestic musical use at both maximal and minimal loudness levels, both aspire to equally ideal acoustic reproduction, and both cost the end consumer the same amount. In the end it will be clear that a minimalistic design of a singular full-range driver has difficult challenges to perfect, but the benefits clearly outweigh the negatives.

A perfectly designed loudspeaker by today's standards would have a flat frequency response, inaudible distortion levels, and approaches an omni-directional radiation pattern from a point source location. The reason why multi-way systems have dominated the market since their introduction is because “frequency response is the single most important aspect of the performance of any audio device. If it is wrong, nothing else matters” (Toole, 2008, p. 372). By separating the entire frequency response between two or three drivers, each one can be designed to handle a simpler range of physical demands. A tweeter with a small light diaphragm can move at 20,000Hz quite easily, while a large

strong woofer can displace the volume of air required for long 100Hz waves of compression and rarefaction much easier. Getting a single speaker to be capable of both air displacement at low frequencies and quick response at high frequencies of the audio spectrum is quite challenging, in early years this was just not possible and thus multi-way systems were truly a necessity for the reproduction of the entire human hearing frequency range. Lately, many companies have been quite successful in creating single drivers that are capable of reproducing frequencies spanning 40Hz to 30KHz and beyond (human hearing stops around 20KHz). However, the design of a perfect crossover – the element which separates the initial signal between the various drivers – has been and continues to be, the subject of much research, debate, development, and controversy.

Crossovers are divided into three categories of classification: passive filters, active filters, or digital filters. Active and digital crossovers both are placed before the amplifiers and thus require an amplifier for each driver in the cabinet. This added cost and setup difficulty may be why passive crossovers are by far the most common in use today. Unfortunately, passive electronic components are not precise enough nor stable enough over the long-term life of a speaker for an engineer's filter response designs to remain stable. Furthermore, the component parts of crossovers are nonlinear in response levels across the acoustic spectrum – particularly, as Korhola observes, under the varying load of a dynamic loudspeaker (Korhola, 2008). This means that when driven with a loud signal a slightly different response curve will be generated from a filter than when the same filter is driven with a quiet signal level. Sigfried Linkwitz (co-inventor of the Linkwitz-Riley filter design) states,

The only excuse for passive crossovers is their low cost. Their behavior changes with the signal level dependent dynamics of the drivers. They block the power amplifier from taking maximum control over the voice coil motion. They are a waste of time, if accuracy of reproduction is the goal (Linkwitz, 2009, “Crossovers”).

Many researchers agree with Linkwitz, Korhola goes on to describe that active and digital crossovers are better, but also fraught with acoustic nightmares and danger zones (Korhola, 2008). One

commonly mentioned culprit is phase shift. Though the importance of phase-accurate reproduction is a debated subject, it is clearly detectable in blind tests and filters steep enough for crossover designs almost always induce phase shift. Furthermore, when multi-way systems play a crossover frequency Toole points out that the multiple point-source locations will inevitably be at least slightly out of phase from one another in the vast majority of listening positions (Toole, 2008). Another clear culprit is the filter topology design itself. Linkwitz points out that

The typical parallel arrangement of crossover filters yields a flat summed frequency response only if the constituent highpass and lowpass filter responses are unaffected by adjacent crossover filter sections.... This parallel filter topology is popular. It is correct for 2-way systems. It usually works for 3-ways when the two xo frequencies are more than a decade apart. It is not appropriate for 4-way systems (Linkwitz, 2012, "Crossover topology issues").

Even with the precise and tuned control of a well engineered crossover, a multi-way system under heavy dynamic load will inevitably experience thermal stress which raises the electrical resistance of the devices, and changes many fundamental Thiele-Small parameters (measurements that are used in the design of loudspeakers), as discussed by Colloms. This then creates inevitable gaps or peaks at the crossover frequencies, or sensitivity imbalances of the driver's entire range due to the altered resistance characteristics (Colloms, 1997). Clearly many challenges exist in a system of multi-way drivers and crossover filters – none of which lend themselves to simple solutions.

These problems aside, there are many reasons why a multi-way speaker system is a useful design. Beyond the previously mentioned physical demands of various ranges, these benefits primarily involve various modulation distortions being reduced when drivers operate on a limited frequency range (Colloms, 1997). Both intermodulation distortion (or doppler distortion), and cone breakup modes are significantly more prominent when a single diaphragm is responsible for a wide range of frequency reproduction. Toole notes, as early as 1936 it was observed that these two issues are directly behind frequency response in a ranking of audibility of problems in audio reproduction equipment (Toole, 1986). These issues have been enough reason to keep professionals away from single-speaker

systems for years, but with advancements in technology it may soon be possible to compensate for or prevent these modulation distortions.

Doppler distortion is the effect of modulating a high frequency with a lower frequency. When two or more frequencies are reproduced by a speaker, for example 4000Hz and 86Hz the lower frequency will modulate the higher frequency (increase it as the cone moves forward and decrease it as the cone moves back) and one or more inharmonic sideband frequencies will be audible at multiples of $\pm 86\text{Hz}$ from 4000Hz with amplitudes that follow a Bessel function (4086Hz, 4172Hz, 3914Hz, etc...). This follows the frequency modulation theory explained by Roads. The larger the difference between the frequencies, the greater the number of sidebands that will be created and the greater the collective amplitude. Thus, when a single speaker is reproducing a very wide range of the audio spectrum a much greater percentage of doppler distortion is inherently introduced (Roads, 1996). In the reproduction of real music rather than two simple sine waves, thousands of frequencies are reproduced at once and thus a very wide band of doppler distortion is introduced to the acoustic signal.

Thankfully, this phenomenon behaves in a predictable and mathematically describable behavior. Because of this, Elliot observes that it is entirely possible to correct this phenomenon with a built-in DSP chip (Elliott, 2004, "Doppler Distortion in Loudspeakers"). The algorithm topology would be to run a Fast Fourier Transform on the incoming signal, put the resulting sinusoidal components of the signal through an appropriately modeled frequency modulation algorithm to find the anticipated synthesized distortions, then synthesize the exact inverse (180 degrees out of phase) signal to eliminate the distortion. Such a DSP chip is not yet being produced, but it is no-doubt within our technological capabilities to create in the near future. Furthermore, studies by Roy Allison and Edgar Villchur, among other voices, have called into question the actual audibility of doppler distortion in acoustic reproduction. They finally concluded that – much like phase – it is a minimally audible phenomenon under strict test conditions (Allison & Villchur, 1980). Thirty years later, doppler and other nonlinear

distortion is still a regularly studied phenomenon with regularly inconsistent subjective results (Henin, Maio de Santis, 2007) leading us to continue wondering how much its effects matter. Wolfgang Klippel (founder of the famous loudspeaker measurement products company) states, “The easiest way to avoid [doppler] distortion is to a multi-way system with sufficiently low crossover frequency between the woofer and tweeter system.” (Klippel, 2005, p. 17).

Another form of speaker error that is greatly increased with full-range drivers is cone breakup. This is when the produced frequencies excite the natural resonance modes of the speaker diaphragm itself and an extremely uneven frequency response results – particularly across the time domain as the resonant energy settles. As Larsen explains, two main factors determine the natural resonance modes of the diaphragm: material properties, and diaphragm geometry. The modal properties of small tweeters are geometrically small enough that the resonances are above human hearing ranges, while common woofer resonances don't begin until the 3KHz region – past the normal crossover frequency for the tweeter range (Larsen, 2003). The problem does become an issue when a single diaphragm is used to produce a wide range of frequencies because the geometry of the cone must be large enough to move a substantial amount of air while also being light enough to quickly respond to high frequency sounds. This large light diaphragm is obviously prone to cone breakup, but full-range driver designers have been adopting various techniques to combat these problems. Colloms observes that one common design utilizes a hyperbolic curve geometry so that the center of the diaphragm is stiffer and radiates higher frequencies while the whole diaphragm is employed to radiate lower frequencies (Colloms, 1997). It is foreseeable that the era of micro and nano technology advances will give speaker designers the ability to use lightweight materials of exceptional stiffness as to eliminate cone breakup in the audible range. There have already been a number of patents filed by Chen, Fan, Jiang and Xiao, among others, in the U.S. Patents office hoping to secure rights to this emerging nano materials technology in loudspeaker designs (Chen, Fan, Jiang & Xiao, 2012).

Future technologies aside, the entire concept of acoustic reproduction is that of point-source sonic reproduction. The origin of most acoustic sources come from a single point in space or at least the wavefronts can be described as coming from a central point-source and the general method of audio recording is through a single microphone point source (panned to the appropriate stereo position). A single driver reproduction method can readily reproduce this (at least in one hemispherical direction). A multi-way system with separate reproduction areas for various frequencies is clearly going to have multiple point-sources for different areas of the frequency spectrum. At first these spaces of five inches might seem negligible to our ability to locate sound – particularly in the vertical dimension – however, these gaps end up changing various reflected sound path lengths to the listener. This confuses our intricate sense of sound localization and spatial recognition. Anyone who has heard a full-range speaker can easily attest to the clarity of the point-source localization.

One particular area where single driver speakers unanimously excel at is in the vocal range. The vast majority of crossover points exist around the 1KHz-4KHz range. This range, as Toole points out, corresponds directly to the vocal range of human speech and thus also happens to be the frequency range human ears have evolved to hear the subtlest differences in (Toole, 2008). The phase changes, inaccurate frequency response contours, and fuzzy localization at multi-way systems crossover point wreak havoc on the delicate subtlety of vocal reproduction. This range however, is right in the linear response region of full-range drivers and thus, they are unquestionably perceived as being superior over this sensitive region of our hearing.

Neither full-range nor multi-way speaker systems are without their difficulties and challenges. Despite the full-range driver having shortcomings, it is evident that most of these issues can be resolved in the near future or with significant expense today. Multi-way speakers have long been accepted as the industry standard for quality sound, but as technology gets closer to a convincing full-range speaker with less cone breakup, and possibly doppler distortion compensation, the defects in

multi-way designs may be overwhelming. Not only will the cost of producing only one driver for each channel reduce hi-fi entry costs to the consumer, it will also bring better hi-fi to the entire market. As it stands the market for various audio devices is demanding smaller and more compact designs on a regular basis – many already switching to full-range drivers despite the current shortcomings. If the hi-fi industry focused its work primarily on making advances in this area the average sound quality of audio reproductions in our personal lives would significantly improve. It is this fork of speaker development that not only holds the most promise for the median audio experience but also the outliers at the forefront of identical reproduction.

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