The Art of Speaker Design

This chapter opens with a quick trip through the future, the past, and the challenges of designing speaker systems in the here and now. These challenges can be met in many different ways, resulting in many different schools of speaker design. Due to the limitations of the art, no one school can "have it all," despite advertising claims to the contrary. At a more detailed level, the designer has to examine the sonic character of different types of direct-radiator driver, and know the advantages and limitations of each type.

Even designers who profess an agnostic, specification-driven approach make an esthetic decision when they decide which group of specifications to optimize. At every point, from overall system design to subtle points of cabinet construction, esthetic preference merges invisibly with engineering decisions.

The Future

If you relax and take a mental journey to the 22nd Century, it is easy to imagine the perfect loudspeaker. It would made of an immense number of tiny point sources that would create a true acoustic wavefront (or soundfield). Resonances due to massive drivers and cabinets would be a thing of the distant past. A host of distortions (harmonic, intermodulation, crossmodulation, frequency, phase, and group delay) would be utterly absent ... the sound would be literally as clear as air itself.

This perfect loudspeaker would be made of millions of microscopic coherent light and sound emitters, integrated with signal processing circuits all operating in parallel. (Similar in principle to present-day military phased-array radars, with tens of thousands of tiny antennas with individual electronics subsystems.)

It would be "grown" by nanotechnology and operate at the molecular level, appearing simply as a transparent film when not in operation. Let your imagination roam free ... this device also has access to all sounds and images ever recorded, and an instantaneous link to billions of similar devices. The primitive 20th Century technologies of telephones, movies, radio, television, hi-fi stereo, and the World Wide Web converge into an apparently simple technology that is transparent and invisible.

The Past

Contemporary speakers, for all of their faults, are better than most speakers of the Fifties. Very few "hi-fi nuts" had full-size Altec "Voice of the Theatre" A-7 systems, Bozak B-305's, 15" Tannoys, or Klipschorns. The typical enthusiast had to endure University, Jensen, or Electro-Voice 12" coaxial drivers in large resonant plywood boxes with a single layer of fiberglass on the rear wall. A large cutout served as the vent, resulting in boomy, resonant boxes tuned much too high, with a 6 to 12 dB peak in the 80 to 150 Hz region. (Have you ever heard a restored jukebox?)

The coax, or worse, triax drivers went into paper cone breakup at 300 Hz and above, cavity resonances (due to the horn element mounted in the cone driver) at 800 Hz and above, horn breakup throughout the working range of the short horn, and phenolic diaphragm breakup at 8 kHz and above. A "good" driver of this type usually had a plus/minus tolerance of 4 to 8 dB, and it took a lot of judicious pen damping to get it to measure that well.

It wasn’t for nothing that early hi-fi systems acquired a "boom-and-tweet" reputation. The sound quality was closer to an old neighborhood theatre, or amusement park skating rink, than a modern speaker. The tube electronics helped sweeten much of the coarseness, but they couldn't rescue the really bad loudspeakers of the day. True, the first-generation Quad, the RCA LC-1A, the Tannoy, and the Lowther compare well with modern systems ... but they were rare, and *very* expensive, at the time. How expensive? The classic speakers cost as much as a new Volkswagen or the down payment on a house!

Peering through the looking-glass of time, we can see that the old designers had no consistent way of modeling or predicting the bass response, and the materials available for tweeters were very poor by modern standards. Today, accurate, design-by-the-numbers bass is taken for granted, and modern tweeters really are superb.

Where modern systems fall down is midrange performance, which doesn’t lend itself to the computer design tools that are so convenient in the bass and treble range. The sparkle and dynamism of the best classic speakers is in the midrange, the most important, and yet the most challenging, part of the entire spectrum. Progress in the midrange region has been slow for many reasons. The ear reaches its peak sensitivity this region, drivers are operating at the edge of their frequency range, and the designer has to contend with spectral flatness, polar response, IM distortion, impulse response, cabinet energy storage, diffraction, and crossover polar characteristics all at once.

By the late Sixties, the big 12" and 15" reflex systems were replaced by the AR's, KLH's, Advents, and other small bookshelf speakers of the Sixties and Seventies. The new speakers had 8" woofers with heavy felted cones, small sealed enclosures, phenolic-dome tweeters, minimal crossovers, and low efficiency. By modern standards, they were dull, dull, dull, with mediocre imaging and coarse, low-resolution sound. This was thanks to the minimal crossover, undamped standing-waves in the box, not using a mirror-imaged driver layout, and diffraction problems with the decorative edges of the box, grill frames, and heavy, non-removable grill cloth.

Although the new bookshelf speakers measured flatter using the simple measurement techniques of the day, the wonderful sparkle and verve of the best Fifties designs was lost. It wouldn't be until the late Eighties, with re-introduction of higher efficiencies, new cone materials, and more powerful measurement systems, that the directness and immediacy of the classics would be regained. Between the late Sixties and late Eighties, "accuracy" and "neutrality" were the primary goals.

I find it interesting that first-generation transistor amplifiers like the Dyna 120, Crown DC-300, and Phase Linear 400 used to be favorably compared to the classic vacuum-tube Dyna Stereo 70 and Marantz 9. (Even by J. Gordon Holt's early "Stereophile" magazine!) That tells us a *lot* about the resolving power of the best speakers of this era. Progressive improvements in speaker design over the decades now reveal the actual sonic quality of these first-generation transistor amplifiers as badly flawed, while the "freshened-up" vacuum-tube classics sound as good or better than the most expensive transistor amplifiers made today.

The Present

With the future and past in mind, we can look at the design challenges of today with fresh eyes. Here's a partial list of the problems faced by contemporary designers:

- 2-speaker stereo falls far short of a true acoustic wavefront, producing a phasey, unrealistic image of small size that causes listening fatigue for many people (particularly non-audiophiles). The virtual image is unstable with respect to listener position, spectral energy distribution, and room characteristics.

  Even a simple central mono image has been shown to suffer from deep comb-filter cancellation nulls between 1 kHz and 4 kHz, which is why a solo vocalist sounds different coming from a single mono speaker and a conventional stereo pair. Psychoacoustic research indicates that 2-channel signal sources require a minimum of 3 loudspeakers to faithfully re-create the tonal quality of centrally located sound sources, such as vocalists.

- Large amounts of harmonic, intermodulation, and crossmodulation distortions combine with mechanical driver resonances to concentrate spectral energy at certain frequencies. Driver damping techniques usually improve spectral characteristics (the frequency response curve looks better as a result) but do not provide much improvement for the underlying breakup modes, so the distortions may actually be spread over a much broader frequency range.

  The narrowband nature of resonant distortions in loudspeakers is why a single-frequency THD or IM measurement is useless; it takes an expensive tracking-generator type of measuring system in order to create a usable frequency vs. harmonic distortion graph. These graphs usually show quite different frequency spectra for the 2nd and 3rd harmonics, as well as curves as rough as undersea topographic maps. Asking for the "average distortion" of a speaker is similar to asking about the "average depth" of the Atlantic Ocean.

  The driver diaphragm needs to have a density equal to air and absolutely uniform acceleration over the entire surface at all frequencies if you want to remove all resonant distortions. We are nowhere close to meeting this criterion. As a result, all speakers have tonal colorations ranging from subtle to gross, with some types of colorations present at all times, and other types of colorations appearing only at high or low levels. A reviewer's preferences in music can easily mask the presence of these problems if they listen to music with a relatively simple spectral structure. (Small jazz trios playing at modest levels, for example.)

- Standing-wave resonant energy is stored in drivers (except for "massless" plasmas), cabinets, and in the listening room itself. The unwanted mechanical energy must be quickly discharged in two ways: rigid, low-loss mechanical links to the earth itself (a rigid path from the magnet to stand to floor to ground), and also dissipated as heat energy in high-loss, amorphous materials such as lead, sand, sorbothane, etc. The energy that is not removed is re-radiated as spurious noise from every single mechanical part of the speaker and cabinet, *each* of which has its own individual resonant signature.

In any real speaker system, regardless of operating principle, there are hundreds of standing-wave resonances at any one time, which are released over times ranging from milliseconds to several seconds. These resonances continually overlay the actual structure of the music and alter the tonal color, distort and mask the reverberent qualities of the original recording, and flatten and blur the stereo image.

In speakers that measure "textbook-perfect," this type of "hidden" resonance is the dominant source of coloration. This is also the reason that 1/3 octave pink-noise measurement techniques have fallen out of favor, being replaced by much more revealing techniques such as TDS, FFT, MLS, and others.

• Radiation patterns shift dramatically with frequency, and change sharply at crossover points; in addition, the radiation pattern is further deformed by diffractive re-radiation at every sharp cabinet edge (regardless of cabinet size or type - this includes compact and planar loudspeakers).

Diffraction, which occurs at every sharp cabinet boundary, creates delayed, reverse-phase phantom sources that interfere with the direct sound from the actual driver. These secondary phantom images create significant ripples in the midrange response (up to 6 dB) and create delayed sounds which disrupt the timing cues necessary to perceive stereo images. These dispersion problems are audible as room-dependent colorations, coarse midrange, diffuse stereo, and a "detenting" effect that pulls images in towards the loudspeaker cabinets.

This list only covers *some* of the problems of loudspeaker systems. There are other problems, not as severe, but still quite audible to a skilled listener. These problems occur in *all* loudspeaker types - dynamic direct radiator, horns, ribbons, electromagnetic planar, electrostatic planar, you name it. They all have lots of harmonic and IM distortion concentrated at certain frequencies, they all store and release significant amounts of resonant energy, and they all have frequency-dependent dispersion further degraded by diffractive re-radiation.

This is why I treat claims of "perfection," or of a "major breakthrough," with a big grain of salt. Does the new wonder technology address even one of the serious flaws cited above? Not too often. The *real* story is year after year of a steady and progressive improvement in materials technology coupled with big steps in measurement technology and computer modeling.

Where Do You Start?

With this background, the most important question of all becomes quite simple: What kind of sound do *you* like?

People actually hear the world in quite different ways, and different people assign importance to different qualities of sound. Some audiophiles value tone above all else, treasuring the sound of their favorite instruments or voices; some like a sense of immediacy, directness, and emotional impact; some like the sensation of an immense 3D space; and others like a see-through transparency, a palpable *"you are there"* quality.

Since all speakers have serious flaws in the absolute sense, it's up to *you* to pick and choose what you want the speaker to do, and how you're going to accomplish that goal. "Perfect Sound Forever" is a silly marketing slogan, not a realistic goal for an artist or an engineer. For one thing, the materials to build anything of the sort simply don't exist. (Unless you've found a way to generate a controllable room-temperature plasma. If you have, you'd better forget about speakers and talk to the Department of Energy first.)

Major Schools of Speaker Design

Since all designers are forced to choose on a subjective (or marketing) basis, there is no single "right" or "wrong" way to design a speaker. If anyone tells you that, it might be interesting to investigate their personal beliefs a little further and see if they worship at a church of religious fundamentalism or the much larger church of "rational-scientific" fundamentalism.

Where do I stand personally? Well, I'm certainly not an audio-fundamentalist, or any other kind of fundamentalist!

I pay attention to spectral flatness, minimizing IM and FM distortion, very low energy storage, and low diffraction. Of course, these objective measurements are only a means to an end. More importantly, I seek an elusive quality I call *"the bloom of life"* ... that "reach-out-and-touch-it" impression of being physically and emotionally present at the performance. For those of you who have never had this experience, I can tell you it really does happen, but only about as often as seeing a perfect double rainbow.

In the section that follows, I'll describe the various paths that designers must choose as they make their way to sonic perfection.

Flat Response (The Objective-Design School)

Most British and Canadian speakers fall in this group. They are characterized by flat frequency response, with the BBC British school assigning the greatest importance to the 2 meter on-axis response curve combined with freedom from delayed resonance, and the NRC Canadian school assigning priority to the frequency response averaged over a forward-facing hemisphere. These design priorities have been arrived at by BBC broadcast professionals and NRC listening panels respectively.

This school of design is most closely identified with an "objective" engineering-oriented philosophy. Not by accident, engineers with masters and doctorates in acoustics tend to design speakers using this philosophy. These folks are *not* going to be sympathetic to exotic wires, resistors, capacitors, the directly-heated triode mystique, or anything not audible on a repeatable basis to a double-blind listening panel.

D.E.L. Shorter of the BBC was the first to accurately measure and identify sources of driver and cabinet resonances in the late 1950's, and many British speakers continue excel in this area as a lasting legacy of the BBC philosophy. Since resonances may be audible as far as 20 dB below a conventional sine-wave response curve, the BBC became the first organization to identify *and measure* colorations that were completely invisible on conventional swept sine-wave measurement systems.

It took American designers 20 years to acknowledge the importance of these "hidden" colorations; the real breakthrough on this side of the pond happened when Richard Heyser invented the Time-Delay Spectrometry system in the early 1970's, first embodied in the Techron TEF test unit. Ten years later, Lipshitz and Vanderkooy invented the "Maximum Length Sequence System Analyzer," which was commercialized by DRA Labs as a one-piece board that could fit into any standard PC. In the space of thirty years, measurement of delayed resonance went from a special-purpose instrument used only within the BBC, a dedicated $150,000 HP FFT minicomputer used by KEF, a $12,000 TEF unit made by Crown, to a $3,500 MLSSA board that plugs in to any PC.

At the time of writing, the MLSSA remains the time-and-frequency measurement tool of choice for major manufacturers. If your are primarily interested in frequency response, and don't care about interpreting the arcana of step responses and waterfall graphs, the $1000 LMS unit is a better choice. The LMS is widely used for quality control in production, since it is easy to set up with "go/no go" limits on frequency response. Another interesting unit is the CLIO, which appears to have similar time-and-frequency performance to the MLSSA, with the additional bonus of 16-bit accuracy and a $1600 price-tag (which includes microphone).

In the last two years, software packages that utilize the top grade of PC sound cards have become available, reducing costs below $600. If you're curious about these hardware/software packages, refer to the ads in the latest issue of "AudioXpress" magazine. The PC sound-card measurement field is changing so quickly that anything I put in here will be obsolete by the time it's printed. The one "gotcha" with most sound cards is a maximum sampling frequency of 44.1 kHz; to accurately measure the impulse response of a tweeter, the anti-alias filter preceding the digital converter must have a relatively gentle slope (Bessel or Butterworth), and this dictates in turn a sampling frequency of 96kHz or higher. If the sound-card vendor is evasive about the maximum sampling frequency or the slope of anti-alias filter, don't buy it.

Moving on to the difficult area of crossover design, objective-school designers usually prefer 4th-order Linkwitz-Riley networks, which offer the flattest, most accurate response curve and the best control of out-of-band IM distortion (at the expense of pulse distortion and overshoot).

Laurie Fincham of KEF deserves credit for pioneering the use of a computer to accurately model the combined electroacoustic behaviour of the driver and the crossover network, allowing accurate synthesis and optimization of acoustical 2nd, 3rd, and 4th-order rolloffs. Prior to Fincham's work, crossover design was a matter of "bending" standard textbook crossovers to get a rough approximation of the desired acoustic characteristic. After Laurie Fincham published his technique, it became a simple matter of deciding the network topology and the desired "target slope," and letting a computer do the cut-and-try guesswork for you.

Of course, back in the early 1970's when this was pioneered, a "computer" meant a dedicated HP minicomputer coming in at $150,000 and a full-time Fortran programmer to punch the card decks and run the thing. (That's what I saw at KEF when I visited them in 1974.) Today, this crossover optimizing technique is now available at far less cost by using a 486 or Pentium PC with ready-to-run programs such as XOPT, CALSOD, LEAP, and others. As a result of the dramatic cost reduction in both crossover optimization and powerful measurement systems, contemporary speaker designers are expected to be well-versed in the use of PC-based tools *regardless* of their design philosophy.

Objective-school designers have until recently ignored pulse response, diffraction control, and those fuzzy subjective areas such as capacitor, inductor, and wire quality. In contrast, research is focused on steadily improving driver quality, cabinet resonance control, and precise pair-matching in production.

Pulse Coherent Dynamics

Dunlavy, Thiele, Spica, and Vandersteen speakers fall in this group. The designer takes extensive steps to control diffraction, offsets the drivers for a coherent arrival pattern, and usually employs a first-order (6 dB/Oct) crossover. Some, such as Spica, may use 3rd (18 dB/Oct) or 4th order (24 dB/Oct) Gaussian or Bessel crossovers.

This is the only type of direct-radiator speaker to offer accurate pulse reproduction, sometimes even outperforming electrostats or ribbon loudspeakers. However, the audibility of phase, and pulse distortion, is quite controversial in the audio engineering community, with the more conservative engineers feeling it is a waste of time and money to ensure accurate pulse reproduction.

In a typical pulse-coherent design, the drivers are asked to be well controlled 2 or more octaves beyond their normal operating ranges, so power-handling and IM distortion are inevitably compromised. Expensive drivers are required to partially overcome this problem, along with accurate resonance correction in the crossover. Controlling the radiation pattern with first-order crossovers and offset drivers is also difficult; as a result, speakers of this type may sound quite different sitting, standing up, and off-axis.

One design technique that I feel is a serious mistake is burying the mid or treble driver in a felt-lined cavity in order to time-align all of the drivers within a conventional speaker box. My experience with wool felt is that it works quite well damping the inside of the cabinet, but expecting it to absorb 100% of a broadband spectrum is silly. No known absorber has 100% absorption across the spectrum; the best you can hope for 20 to 30dB of attenuation in the desired band ... and that takes a lot of different materials in a composite assembly several inches thick. When I damp the floor bounce for MLSSA's benefit, I have to use 2 *feet* of miscellaneous fluffy materials in order to get that one single reflection attenuated by 30 dB. Imagine how little effect 1/2" of felt is going to have.

Placing a driver at the back of a hard-surfaced cavity is going to give you a very obvious "honk" coloration similar to cupping your hands around your mouth when you speak. Well, lining the cavity with thick wool felt helps a bit (and makes MLSSA happier), but the "honk" sound is still there if you listen for it. Not only that, but getting the felt right next to the driver mass-loads the diaphragm, reduces the efficiency, and degrades the transient response as well. All this hassle just to get pretty square waves? Nuts! If you *must* offset the drivers, do yourself a favor and use separate boxes, baffles, or whatever for the mid and high-frequency units.

When done right, pulse-accurate dynamic systems can sound as open and "free" as electrostatic speakers, particularly if it is a low-diffraction design. The downside can be limited dynamic range for the tweeter or midrange and a complex polar pattern with a narrow sweet spot.

(Actually, I'm not as prejudiced against these systems as it might sound. My last speaker for Audionics was the LO-2, and it was a pulse coherent sub/satellite system exhibited in the 1979 Winter CES. The satellite used a 6.5" Audax Bextrene midbass, a 1.25" Audax soft-dome tweeter, a 2nd-order Bessel crossover, and yes, it could reproduce recognizable square waves. They were part of my main system right up through 1993, when they were replaced by the Ariel transmission-lines.)

Minimalists

Some Italian, Scandinavian, English, and American speakers fall in this group. The crossover is very simple, sometimes reaching the extreme of one capacitor! Drivers and crossover components are of the very highest quality, along with exotic wire and cabinet materials.

Measurements usually play a minor role in the development of this type of speaker. Since this design philosophy leaves driver resonances uncorrected and accepts the resulting frequency and pulse response aberrations produced by the minimal crossover, compatibility will probably depend strongly on the sonic flavor of the rest of the audio chain.

Even though few designers are all-out minimalists, the "parts quality really matters" thinking of this group has influenced much of the rest of the industry. As far as I know, no reputable high-end speaker manufacturer uses electrolytic capacitors in the crossover these days, and Mylar isn't too common. This is a significant change from the Seventies, when even the most technically advanced speakers of the day routinely used crossover parts that we would now consider to be no better than floor-sweepings. Twenty years ago, the focus was almost exclusively on drivers, design technique, and cabinet construction. Today, careful designers examine all parts of the system, right down to the fasteners used to mount the drivers and the type of plating used on the input jacks.

Now if you really want to get down to basics, you can't get any simpler than a full-range single-driver system. No crossover, no worry about shifting radiation patterns, and no phase distortion from the all-pass characteristic of the crossover. The "gotcha" is the extraordinary difficulty of building a full-range driver that sounds good. The requirement for genuine bass output combined with extended high-frequency response has only been solved by the Lowther company, which has been making a series of 6" full-range drivers first designed by P.G.A.H. Voight more than 50 years ago.

When you see a Lowther driver for the first time you are struck by its unusual appearance; the spiral grooves embossed onto the stiff white paper cone (I've been told this is waxed cartridge paper used for making gun shells), the small whizzer cone (Voight patented the whizzer cone in the Thirties), the extremely large magnet (Alnico is available and is considered the best), and the very short travel (less than 1mm). What you can't see are the magnetically saturated pole-pieces, and a short gap with extremely close tolerances. These combine to provide an extremely high BL product, in the same range as compression drivers for studio-monitor horns.

In a dynamic driver, a high *"BL-product"* means a strong magnetic field in the gap (the *B*) and a long helical voice coil immersed in the field (the *L*). Dynamics with a high BL-product provide the tightest amplifier-speaker coupling, which is why they are sensitive to amplifier damping factor and wire resistance. The opposite extreme would be a magnetic-planar, which has a very low BL product. So when you hear a difference between a Lowther and the magnetic-planars, that's one of things you're hearing.

The Lowther driver is designed from the outset as a horn driver, and indeed, cannot and should not be used in conventional enclosures due to its rising high-frequency response and limited voice coil travel. Those of you who fool around with Thiel/Small equations are probably aware that as the magnet of a driver gets more powerful and the Qt drops, the bass response drops away as well. The limiting case is the Lowther driver, which has such a powerful magnet system that the response is tilted upwards throughout the entire frequency spectrum. This where horn loading comes in; it provides the greatest efficiency gain at low frequencies, compensating for the rising response at the same time it reduces excursion by stiffening the air load. Think of the Lowther as a 6" big brother to a high-quality 2" compression horn driver and you'll get the idea.

With a Lowther, the entire design process boils down to selecting and building an enclosure. This is a bit more complex than it might first appear; Lowther-club enthusiasts have been designing all kinds of enclosures since the Fifties, and there are hundreds of designs and variants out there. Horn enclosures are also the most difficult of any type to build, since they have very complex shapes internally, and must be made very accurately from rigid materials. (3/4" Baltic birch plywood is a good starting point; forget about MDF.)

Horns and High-Efficiency Systems

As mentioned in the previous chapter, speaker systems in the mid-Fifties were quite efficient by modern standards. The contemporary audiophile favorites such as electrostats, planars, and minimonitors have efficiencies around 82dB/metre with 1 watt input (about 0.1%), while the most popular hi-fi speakers of the Fifties had efficiencies around 92 to 96dB/metre (about 2%). The bigger and more prestigious theatre systems adapted for home use had efficiencies as high as 102dB/metre (10%) ... the same as contemporary PA systems and studio monitors.

What happened? It was an article of faith in the Fifties that the best speakers were the most efficient. Hi-Fi fans were well aware that Western Electric, Altec, and RCA theatre speakers represented the most advanced speaker engineering available. If you wanted proof, you could go to the movie theatre and become immersed in "This Is Cinerama," "Ben-Hur," or "20,000 Leagues Under The Sea."

This faith in efficiency as a virtue in itself was weakened by the introduction of the AR-1, the first small-box acoustic-suspension speaker that was correctly designed. Although it was ten times less efficient than the speakers it challenged, it really did go down to 30Hz with no boom, and it was compact as well! It took a while for amplifier power to catch up to the demands of the acoustic-suspension speaker, but by the time of the advent of stereo in the late Fifties, 60 watt/channel amplifiers were coming on the market. Speaker designers were now willing to trade off a little efficiency here and there in return for better damping and control. Now that the long-sought goal of an measurably "accurate" speaker was actually coming within reach, many designers were willing to try better-damped and less-efficient materials to get there.

This is when the "West Coast Sound" versus "East Coast Sound" catfight really got rolling. The Westerners were represented by JBL, Altec, and Cerwin-Vega, and the Easterners by AR, KLH, and Advent. Over the course of the Sixties, the Westerners ended up building smaller and smaller speakers that tossed away the efficiency and dynamics of the good theatre systems, but copied and quite deliberately exaggerated the bass boom and horn colorations of yesteryear. The most famous example of this marketing philosophy was the very successful JBL L100, a beautiful-looking bookshelf speaker with a bright-orange sculptured foam grille. It really looked great until you had to listen to it.

The Eastern school, in reaction to the shrillness of the early transistor amplifiers and the aggressiveness of the West Coast Sound, gradually made their speakers more and more muffled. They measured flat, unlike the Westerners, but no attention was paid to driver distortion, crossover refinement, or reducing cabinet coloration. If you open up one of these today, you'll see one very cheap electrolytic capacitor connected to the tweeter, a loose wad of fiberglass, no cabinet bracing of any kind, and a thick grill cloth stapled to a massive overhanging frame. By the early Seventies, American audiophiles were getting tired of the crude design and mediocre build quality of both the West and East Coast schools, and started to looking across the Atlantic for something with a bit more class. The British were happy to oblige.

In the 1970's, the UK was at the forefront of world research in loudspeakers, pioneering new materials like Bextrene, computer modeling crossovers, and using FFT techniques to track down resonances in the drivers and the cabinets. By now amplifier power had soared into the 120 watt region, so designers still felt free to discard a dB here and there in order to control resonances and smooth the response.

The nadir of efficiency was reached in the early Seventies, and best exemplified by the BBC-designed LS 3/5A. This is a wonderfully smooth and articulate speaker that transcended the West Coast vs. East Coast Sound debate, and it immediately won acceptance amongst audiophiles worldwide.

All was not sweetness and light, though. The heavily-damped 5" Bextrene cone of the B110 dragged the efficiency of this petite wonder all the way down to 82dB/metre. This speaker actually did require a 200 watt amplifier to "open up" and play music. More than one satisfied LS 3/5a owner actually had amplifiers that were bigger and heavier than their speakers! Still, it was a good choice in its day; amplifiers were finally sounding better, and you no longer had to choose between sound that was piercingly bright or dull and muffled.

A couple of years later, the KEF 104a was introduced. KEF refined the mid and tweeter in the LS 3/5a, designed a new bass driver to succeed the B139, and came out with the first speaker to use a computer-optimized Linkwitz-Riley crossover. Although the efficiency was still no higher than the LS 3/5a, the 104a set new standards for naturalness, clarity, and image quality (a direct result of the advanced crossover).

The countertrend towards higher efficiency and lower power started in the late 1970's, with more efficient cone materials like polypropylene and lower-powered transistor amplifiers (designed in accordance with Matti Otala's non-slewing criteria). By the late 1980's, a tube revival was well underway, and speaker designers had a wide variety of materials to choose from, with new-and-improved paper cones, polypropylene, Kevlar, and carbon fiber.

The new materials did not require any external damping compounds (unlike Bextrene), and relied on internal self-damping within the cone material itself. Holographic imaging and computer modeling systems led to a series of gradual refinements in cone materials, tweeter diaphragms, and greatly improved design of the magnetic gap with ventilated pole-pieces for bass, midrange, and treble drivers. At the time of this writing, the best direct-radiator drivers from Scan-Speak, Dynaudio, Audax, and Focal now have efficiencies between 89 and 94 dB/metre, representing a fourfold gain over the Seventies.

Joe Robert's "Sound Practices" magazine had a major effect on the North American market by exposing it to schools of audio design from Japan, Italy, and France. The overseas ultra-fi fans didn't have sour memories of the "West Coast Sound" marketing disaster, and continued to hold the classic high-efficiency theatre speakers in high regard.

Outside of the Anglo-American orbit, the design philosophies of "old" Western Electric theatre speakers, Altec and JBL studio-monitor systems, and P.G.A.H. Voight's Tractrix horns are still taken quite seriously. The appeal isn't nostalgia; brand-new drivers and horns made from exotic materials appear on the market at prices that would astound Western audiophiles. These alternate-paradigm speakers work especially well with flea-power amplifiers using direct-heated single-ended triodes; a 3 watt 2A3 amplifier simply doesn't work with room-sized electrostats or planars, but works beautifully with a 104dB efficient all-horn system.

To those who think amplifiers have already reached near-perfection (almost all of the AES establishment and home-theatre vendors), this embracing of archaic "foreign" technologies looks like some kind of bizarre joke. The slick high-end magazines explain away the horn/triode phenomenon as retro-chic, just another trendy example of mythologizing the past.

The flip side of this coin is the fact that the most articulate horn/triode advocates have *already* owned, and discarded, mainstream audiophile systems. As a fairly mainstream speaker designer myself, I can attest that raising the efficiency of conventional direct-radiators is *most certainly* worthwhile ... you get a significant improvement in clarity, immediacy, and naturalness, *and* your choice of amplifier opens up to much more interesting technologies.

From a technical standpoint, horn-loaded drivers typically have very low THD, IM, and FM distortion, uneven frequency response, reflections in the time domain, and very sharp cutoff characteristics at both ends of the frequency range. From the viewpoint of mainstream high-end designers, horns are beset by serious problems with impulse response, diffraction, and smooth dispersion.

The root of these problems, especially with cheaper PA-style horns, is the acoustic reflection from the edge of the horn-mouth. When a sound wave moves across a sharp boundary, it diffracts and re-radiates in all directions, like a separate driver located at the point of the reflection. The reflected wave from the horn-mouth then bounces back into the throat, which typically has a hard phase plug or a driver with a stiff cone. After it strikes that, it reflects right back outward again ... this succession of reflections is called a series reflection, and it is far more audible than the small ripples in the frequency response might indicate.

Although the frequency response doesn't really indicate the full impact of the reflections, they show up in the impulse response or 3D waterfall display. (This is most clearly seen if the horn driver is measured without a crossover.) Inexpensive PA horns that are too short suffer most severely from this problem, and have the grossest "horn coloration" as a result.

There are solutions for this problem that work pretty well. If you can afford to lose 1 or 2dB or efficiency, you can line the inside of the horn with 1/8" wool felt. 1 to 3 inches extending from the lip of the mouth going inward will do the trick. The further you go back towards the throat, the better the damping, but if you overdo it, the bass response of the horn will start to droop, along with the efficiency. Think of it being like tweaking the VTA on your cartridge and you'll be heading in the right direction. Of course, if you have access to a MLSSA or similar FFT system, you can adjust the impulse response to taste, as well as compensate the crossover accordingly. (Note: if you're modifying a commercial horn, don't forget to remove the wire mesh bug screen in the throat. The wire mesh creates a very unpleasant gritty harshness at levels above 90dB, and is only required for severe outdoor environments.)

The best solution is to eliminate the mouth reflection entirely. This has already been done with the Tractrix horn profile, invented by P.G.A.H. Voight in the late Twenties!

The Tractrix still has a sharp edge at the horn mouth, but the horn wall has already curved through 90 degrees before the sound hits the boundary. The reflected sound then has the difficult task of curving *back* through that 90 degree curve before it can strike the phase plug. Therefore ... no standing wave, only one modest reflection, and very little of the "horn sound" if the compression driver is correctly designed.

(Note: there are rectangular horns on the market that are Tractrix-profile in only one dimension; since the reflection is still an unresolved problem on two of the mouth edges, most of the benefit of the Tractrix profile is lost.)

Building a square or circular mouth Tractrix horn is no simple exercise, and I defer to "Speaker Builder" and "Sound Practices" magazines for the complex procedure on how to make these things. If you're getting the impression that doing justice to horns is a complex and expensive exercise, you're absolutely right.

For example, the best 2" compression drivers for a 500Hz to 22kHz horn are made by JBL and TAD for studio-monitor use, and they cost $800 each, not including the horn! Compare that to a top-of-the-line Scan-Speak driver at $120, and the difference in parts cost becomes obvious. Yes, you can get entry-level PA horns for about $80, but you *really* get what you pay for in the prosound business. Don't expect a grunge-band PA driver to sound like a JBL 2" titanium-diaphragm studio-monitor driver. They may look the same on the outside, but they're *not* the same on the inside.

With horns, the difference in quality between the best and "bad" is really large, and much more obvious than the differences between audiophile speakers. Not only that, the best ones are seriously expensive, requiring machined Alnico magnets, diaphragms made from exotic metals, and horns with compound curves made to exacting dimensional tolerances. It's not a technology that lends itself to cost reduction. On the bottom end of the market, we get low-grade PA speakers, which require a lot of careful modification before than can get anywhere near the "high fidelity" appellation.

Despite all the challenges from the mainstream audio-press establishment, I expect this market to grow in the years to come. It will probably be dominated by well-capitalized companies that can afford to spend a million dollars or more for tooling and start-up costs. JBL, Altec, and Tannoy already sell hand-crafted domestic versions of professional studio monitors for the Japanese high-end market; if the US market grows, they will almost certainly sell the same models here.

Electrostatics

A small group of English, American, and Japanese firms handcraft electrostatic panels, which I have to admit are long-time favorites of mine. A well-designed electrostat offers the most linear and completely uniform diaphragm motion of any class of loudspeaker (and very low IM distortion as a result), as well as the potential for the best pulse response. The original Quad ESL57 is the most famous example of a speaker decades ahead of its time. The old Quad still sounds and measures very well indeed ... if your University research project requires real square waves and very low distortion, the Quad ESL57 will fit the bill.

There are significant problems with electrostatics, though. For starters, we have to contend with: very low efficiency, extremely reactive amplifier load, restricted dynamic range, fragility, limited bass, and a tricky room-sensitive dipolar radiation pattern that becomes quite directive at high frequencies. These problems are *not* easy to solve, particularly the large-panel dispersion, which is not an asset, but a serious problem for stereo imaging.

The original Quad ESL pioneered what is still the most widely used solution, a side-by-side array of vertical panels. It is a 3-way speaker with 6dB/octave crossovers integrated into the step-up transformers. The vertical tweeter strip is on the inside, flanked on the left and right by two midranges, and flanked in turn by two bass panels. The vertical dispersion is mildly improved by the curvature of the panels, while the lateral dispersion is quite narrow due to the side-by-side driver layout. (The resulting listening area for good stereo is about 1 foot across!) To give the original Quad its due, it was designed before the requirements for stereo imaging were known.

The current Quad ESL is a 1-way speaker that uses a complex phased array system (borrowed from radar technology) which approximates a spherical radiator. The new model has rather different sonics, much better image quality, deeper bass, and greater power-handling. Some Quad fans like it, and others prefer the "classic" model, since each design has its strengths and weaknesses. New and different solutions for electrostatics continue to appear every decade or so, as designers contend with the challenge of getting good high-frequency dispersion combined with a large radiating area.

Although electrostatics measure superbly in the midrange, they do not measure "textbook-perfect" over the complete spectrum. All of the electrostats I have checked show moderate resonances below 200 Hz (primary room-diaphragm resonance) and multiple narrow resonances above 8 kHz (from non-homogenous diaphragm motion and standing waves between the HV stators or metal grill-frame assembly). The real claim to fame of electrostatics is the midrange, where freedom from distortion and resonance define the state of the art, and pulse reproduction is good enough for use as a laboratory reference.

In short, utterly wonderful midrange and depth perspective, good-but-not-great at the frequency extremes, reasonable-to-fair stereo imaging, limited dynamic range, low efficiency, and a very reactive load for the amplifier.

Magnetic Planars and Ribbons

Most of these types are made in the USA, represented by Magneplanar (the pioneer), Apogee, Eminent Technology, and others. These fall in two classes:

- **Magnetic-planars**, which are sheets of stretched Mylar or Kapton film with an aluminum "voice coil" either printed or glued on the film.
- **True ribbons**, which use a very thin corrugated aluminum "voice coil" hanging freely like a streamer

in a side-by-side magnetic field.

Magnetic-planars use arrays of magnets on the back side of the film (not too good for IM distortion) or on both sides (much lower distortion, but also creating a small resonant cavity between front and rear magnet pairs). The arrays of magnets provide a somewhat uneven drive field, so the uniformity of diaphragm motion is not in the same class as an electrostat. Then again, HV arcing is not problem, so the magnetic-planars can play much louder than their electrostatic cousins.

Magnetic-planars have a lower BL product than conventional direct-radiators as a result of the much wider pole-to-pole magnet spacing and the shorter length of wire immersed in the magnetic field. Diaphragm damping is mostly provided by the air load, and very little comes from the amplifier. In electrical terms, it is very loosely coupled the amplifier, which is why the impedance curve is resistive (if the BL-product were any higher, you'd see the typical reactive up-and-down impedance curve exhibited by conventional drivers).

Although a resistive load is great for the amplifier, it's not too great for the driver. Drivers are naturally reactive, since all of them are bandpass filters with a bandpass much narrower than the full-range amplifier that drives it. Since the amplifier is driving the band-reject region, a tightly-coupled driver will present a load that looks like a filter ... this is the starting point of Theile/Small theory. The only way to have the amplifier see a resistive load is: design complex pseudo-crossovers that are the inverse (conjugate) of the total speaker load, or use drivers that have very loose magnetic coupling.

The idea that a perfect loudspeaker would present an amplifier with a resistive load is a marketing myth. This myth would only come true if some genius could design a single driver with a working bandwidth of, say, 10Hz to 100kHz. If such a wonder driver were available, who would care about the amplifier load?

Returning to magnetic-planars, the only ways to improve the coupling and raise the efficiency are:

1) Decrease the magnet spacing. This limits the excursion and adds to the requirements for a precise and rigid frame that holds the opposing magnets apart.

2) Increase the number of "turns" by lengthening the path of the wire or aluminum plating on the plastic film. The limit to this approach is adding excessive mass to the diaphragm, which degrades both efficiency and the transient response. Doubling the diaphragm mass cuts the efficiency to one *quarter* of the original value, so most designers go out of their way to prevent adding mass to the radiating surface.

3) Use magnets with higher coercivity. The newest magnets using exotic rare earths may offer significant improvements here. As the magnets get stronger, though, the requirements for a stronger frame also increase.

Magnetic-planars designers confront a series of design challenges that are not too different from the ones presented by electrostats. One has to wrestle with powerful magnet arrays that want to twist on the mounting frame, the other with high-voltage arc-over punching holes in the diaphragm.

Let's move on to a technology that looks superficially similar, but actually is quite different than the preceding magnetic-planars. The freely hanging *true ribbon* is free of the stretched film resonances and obstructing magnets of the planar-magnetic, so it offers outstanding pulse response, uniform drive, and a good approximation of a line source. On the other hand, the impedance is extremely low (a fraction of an ohm) and ribbons are not suitable as a woofers or midrange drivers due to the small radiating area. Most practical ribbons require a matching transformer in order to successfully couple to the amplifier.

Planar speakers, being free of any kind of enclosure, have resonance-free reproduction in the important 100Hz to 1kHz region, resulting in sound quality is usually midway between a good dynamic and an electrostatic, with a genuine freedom from cabinet colorations. (Flexing modes in the supporting frames can be a problem, though.) The large surface area of the panels, their ability to operate at sound levels approaching horns, and the lack of lower-midrange coloration makes the planars a good match for music with a really big sound, such as large-scale symphonies or choral groups.

Like their electrostatic cousins, resonances appear in the 40 to 200Hz region as a result of drum modes on the panels coupling with room modes, so careful room placement is no casual matter. In addition, though, the side-by-side arrangement of the bass, mid, and treble drivers provides a very complex and "lobey" radiation pattern at the crossover frequencies, so the requirements for the best stereo imaging may well conflict with the location that provides the smoothest bass. In short, these speakers work best in a large, symmetric room, with a very powerful amplifier to compensate for the low efficiency and low BL product.

These kinds of speakers aren't my cup of tea, but I know many people who really enjoy the neutral, relaxed type of sound they offer. In addition, a *true* ribbon offers some of the best treble around, surpassed only by the plasma driver.

Plasma Speakers

One day, I'd like to design one of these myself. The "massless" speakers fall into this category ... Ionovac, Magnat, and Plasmatronics (what a name!) They *do* sound exotic, and measure the same. No resonances at all, and accurate pulse and frequency response up to 100kHz or more. Low distortion too ... like a really good amplifier. Actually, the "diaphragms" do have mass. But it's not much. It's the same as the surrounding air, so the acoustic coupling is 1:1. The efficiency is a little difficult to state, though, since the output tubes of the power amplifier are supplying a high voltage that directly modulates a conductive gas with very complex electrical properties.

I first heard the Hill Plasmatronics at the 1979 Winter CES, and I must say I've *never* heard a tweeter that even came close to that one. The exhibitors darkened the room for dramatic effect, and you could see this weird violet glow through the grill cloth that looked for all the world like a gassy triode ... but it was the *tweeter!* Not only did it glow, it pulsed along with the music!

The rest of the speaker, though, was a pretty mundane paper-cone setup in a huge cabinet ... oh well. Even so, the Plasmatronics was a wild thing, a taste of the future, like a SR-71 Blackbird in an airport full of commuter-shuttle 737's. Not too surprisingly, the inventor was a plasma physicist at Los Alamos Labs.

(Talk about being ahead of your time! This was *10 years* before the fall of the Berlin Wall, and Dr. Hill was already thinking of ways to peacefully convert atomic swords into sonic plowshares!)

There are a few little problems with this glimpse of Paradise. Previous generations of plasma speakers, such as the DuKane Ionovac tweeter of the Fifties, used RF heating to ionize air, making it conductive. There's a problem with this. If you ionize air, some of the oxygen molecules (O_2) are stripped apart and then recombine as ozone (O_3). You also get nitrous oxide (NO_2), which is formed by combining nitrogen and oxygen at very high temperatures.

Well, a dose of laughing gas may or may not enhance the listening experience, but ozone really isn't too healthy, since it irritates and burns the mucous membranes and the eyes. The natural home for these highly reactive gases is far up in the ionosphere, not in your living room.

The Hill Plasmatronics avoided the air pollution hazard by having its own built-in supply of helium, which is a noble gas and thus unreactive even when ionized. Helium is also biologically inert, and being much lighter than air, promptly escapes to the upper atmosphere and outer space. Even in the best-insulated houses it will be gone in a matter of seconds, so it is completely safe.

I remember seeing the helium tank, pressure gauges and all, in a special compartment inside the subwoofer enclosure. Imagine cracking a valve and hearing a very faint hiss of helium gas every time you turn on your hi-fi. Oh, I nearly forgot, you had to swap the helium tank for a fresh one every month. Helium is not a renewable resource, and is only found in a few natural gas wells, so it's not as inexpensive as other industrial gases.

There are still some interesting plasma-speakers that haven't been tried yet. For example, one alternative to tanks of helium is a flame speaker (flames are plasmas too), using flammables that release no toxic byproducts of combustion. This leads us to hydrogen and oxygen, preferably generated on-the-spot by splitting water by electrolysis. (You'd water the loudspeaker like a plant!) The hydrogen and oxygen pipes go up to a copper wire mesh (a hemisphere would be the right shape), and the flame is trapped on the surface of the copper mesh.

The system has a computer-controlled power supply that splits the water, monitors the gas flow, and automatically sparks the flame when the correct hydrogen-oxygen ratio is reached. You then polarize the plasma with a high-voltage supply and modulate the flame with a high voltage audio transformer or direct-couple to 211, 845, or 212E plates from the built-in power amplifier. (The flame is modulated in the same way as a conventional electrostatic speaker.)

As far as I know, nobody has *ever* built a complete system like this before. I hereby throw the idea into the public domain, and wait to see if anybody is crazy enough to actually build it.

Don't expect to get UL or VDE safety certification for a "loudspeaker" that mixes hydrogen and oxygen gas, high voltage, distilled water, AC power, an open flame, and a microprocessor, all in a domestic environment. Imagine the reaction of the reaction of the insurance company if they discovered how it works!

Aside from these trivial non-audiophile considerations, the plasma-flame speaker would have truly exemplary performance ... very low distortion, perfect impulse response, and a bandwidth of 100kHz or more. Another benefit of the confined-flame speaker is the "diaphragm" can be as big as you want, limited only by concerns like combustion noise, room heating, and fire hazard. As a compromise, a 6" diameter hemispherical flame front certainly wouldn't be too difficult to build. That would deliver response down to 200Hz or so. It would be lab-standard flat from 200Hz to 100kHz, and no cabinet resonances either.

Just imagine a cold winter's evening with twin pale-blue flames illuminating the copper-mesh hemispheres, the faint hiss of hydrogen & oxygen gas, the quiet murmuring of the water electrolyzer, and a pair of eighteen-inch-high Western Electric 212E direct-heated transmitter tubes to make it all sing. Add a Jacob's Ladder for visual stimulation and the picture is complete; Bride of Frankenstein has an electrifying night over at Nikola Tesla's bachelor pad. Careful with that Zippo lighter, Nick!

[Part Two]

The Family of Direct Radiators

Ahem. I must reluctantly draw the curtain on this depraved scene of electro-motive-force before it proceeds any further, and gently but firmly steer our attention back to the topic at hand.

As we descend from the ethereal realm of charged plasmas, we must once again contend with solid materials ... the same materials that Rice and Kellogg used to build their first direct-radiator cone loudspeaker in 1928. Back then, they made the cone from paper, and paper is *still* used today. We also have new materials that spring from high technology, such as Kevlar, carbon-fiber, ceramic, and impact-forged aluminum, magnesium, and titanium. In the years to come, we can expect new composites, synthetic diamond, ultralow density aerogel-silica glasses, and new types of monocrystalline materials.

The direct-radiator cone has only one task to perform: transform the accelerations of the voice coil into acoustic power over the desired frequency range. To accomplish this deceptively simple task, the driver designer must balance uniformity of motion (rigidity) with freedom from resonance at mid and high frequencies (self-damping). This is the number one sonic tradeoff in all drivers (except the plasmas). There are other problems introduced by cavity resonances and magnetic non-linearities, which are discussed later.

Uniform Motion

Rigidity means accelerations from the voice coil are accurately translated into cone or dome acceleration over the entire driver surface; this translates to ruler-flat frequency response, fast pulse risetime, low IM distortion and a transparent, "see-through" quality to the sound.

Audiophiles usually describe this type of sound as "fast," much to the dismay of measurement-oriented engineers. "How can a woofer *possibly* be fast, since the crossover limits the pulse risetime to a tenth of what any tweeter can do?" This leads to what diplomats discreetly call a "full and frank exchange of views," in other words, a shouting match.

As usual, both sides are right, and both sides are wrong. They're just speaking about different things. The audiophile is unwittingly describing *uniform cone motion*, and it can be indirectly measured by the absence of IM distortion, a flat frequency response in the working range, and good pulse response with a smooth and quick decay signature.

Well, that's great, you might think, just make the cone, or dome, or whatever as rigid as possible. How about a metal, like aluminum, perhaps? That's nice and strong, and it can be formed into nearly any shape.

You can see the direction this is taking. Remember, *bells* are made of metal! Another problem raises its head ... resonance! After all, why does a bell, or any other rigid material, ring so long, for many thousands of cycles?

We need to take a close look at how the mechanical energy gets released (if it didn't get released the bell would ring forever). Well, obviously there are some resistive losses in the bell itself; even in a vacuum the bell will quit ringing after a while. The major loss path is through the air; in effect, the air discharges the stored mechanical energy of the ringing bell. But since there is a very large mismatch between the density of the air and the metal, the coupling is very inefficient, and the bell rings for a long time before all of the energy gets discharged.

Well, guess what? All of these things happen in a speaker cone, too! The cone is much denser than air, resulting in the typically low efficiency of most direct-radiators. (89dB at one metre with 1 watt input corresponds to an absolute efficiency of a mere 0.5%) In addition, the air is so weakly coupled that it doesn't help much with damping the cone (unlike a large-area electrostat or magnetic-planar). We can only look for help from two sources; amplifier damping, which controls the voice-coil, and the intrinsic self-damping properties of the cone and the surround.

Self-Damping

We'd like the amplifier, acting through the voice coil, to *stop* the cone or dome, not have the cone keep playing a tune all by itself. Unfortunately, the voice coil represents only a small portion of the cone area, and the rest of the cone may have almost no self-damping, particularly if it is made of metal, carbon-fiber, or Kevlar. One way to control the problem is to extend a rubber surround partway down the cone, and pay a lot of attention to the damping behavior of the spider and surround materials. (I have heard from several sources that Kurt Mueller of Germany makes rubber surrounds with superior damping qualities.)

At present, though, even the best Kevlar, carbon-fiber, or aluminum cones show at least one high-Q peak at the top of the working range, requiring a sharp crossover, a notch filter, or both to control the peak. Unfortunately, this peak usually falls in a region between 3 and 5 kHz, right where the ear is most sensitive to resonant coloration.

Most audiophiles and magazine reviewers are unaware of the sonic signature of Kevlar or carbon-fiber resonance, misidentifying it as "amplifier sensitivity," "room sensitivity," or other problems that point away from the real culprit. Since few reviewers have auditioned the raw, unmodified sound of commonly-used drivers, they can't evaluate how much "Kevlar sound," or "aluminum sound," remains as a residue in the finished design. It is the task of the designer to skillfully manage the crossover and cabinet profile to minimize the driver coloration. Despite advertising claims or the opinions of nationally famous reviewers, the characteristic signature of a driver can never be removed completely.

When working with rigid-cone drivers, there are some hard choices to make: if you *lower* the crossover frequency to minimize driver coloration, tweeter IM distortion skyrockets, resulting in raspy, distorted high frequencies at mid-to-high listening levels; if you *raise* the crossover frequency to improve the sound of the tweeter, the rigid-driver breakup creeps in, resulting in a forward, aggressive sound at moderate listening levels, and complete breakup at high levels. (Unlike paper cones, Kevlar, metal, and carbon fibers do *not* go into gradual breakup.) With the drivers we have today, the best all-around compromise is a 2nd, 3rd, or 4th-order (12-24dB/Oct.) crossover with an additional notch filter tuned to remove the most significant HF resonance of the midbass driver.

I should add, by the way, that I like Kevlar and aluminum drivers very much ... but no question about it, they are very difficult drivers to work with, with strong resonant signatures that must be controlled acoustically and electrically.

As mentioned above, rigid cones have advantages, but they are difficult to damp completely. An alternative approach is to use a cone material that is made from a highly lossy material (traditionally, this was plastic-doped paper, but this has been supplanted by polypropylene in most modern loudspeakers). The cone then damps itself, progressively losing energy as the impulse from the voice coil spreads outwards across the cone surface. The choice of spider and surround are then much less critical.

This type of material typically measures quite flat and also allows a simple 6dB/Octave crossover; personally, though, I don't care for the sound of many polypropylene drivers, finding them rather vague and blurry-sounding at low-to-medium listening levels. Without access to a B&K swept IM distortion analyzer, I have to resort to guesswork, but I strongly suspect that this type of driver has fairly high IM distortion since it is a soft cone material.

It is quite difficult to make a material that has perfectly linear mechanical attenuation. In the electrical world, we expect resistors to have almost zero distortion. In the mechanical world, though, lossy (soft) materials tend to have weird hysteresis modes, and linear behavior *cannot* be taken for granted. This is the source of the IM distortion in the midband of a driver's frequency range, where the displacement is low, and it is operating in a constant-acceleration regime. In short, it has moderate cone (or dome) flex, but it isn't the all-or-nothing gross breakup that people see in the acoustic holography pictures.

I suspect (without proof) this is the problem for many soft-dome tweeters and midrange domes; the driver is actually flexing throughout the entire frequency range, but the lossy damping material hides this from the instrumentation (but not the ear). To overcome this, the best cone drivers (Scan-Speak, Vifa, and Seas) are actually composites, adding silica, talc, or metal dust to the plastic cone, which significantly improve rigidity without losing the characteristic polypropylene smoothness.

Cavity Resonances

Even though the dust cap in a mid/woofer (or the dome in a tweeter) looks pretty harmless, the space between dustcap and the polepiece of the magnet creates a small high-Q resonant cavity. One example of this was the KEF B110 Bextrene midbass driver dating from the early Seventies (as used in the BBC LS 3/5a).

Although this driver was probably the one of the first high-quality midranges available, it also had a host of problems, such as low efficiency, limited power-handling, a broad one-octave peak centered at 1.5 kHz (corrected by the BBC crossover), and group of 3 very high-Q peaks centered around 4.5 kHz (only slightly attenuated by the BBC third-order crossover). These upper peaks, which reviewers mistakenly ascribed to the tweeter, were also very directional, which is typical of dustcap resonances.

The popular tweeters of the 1970's, including the Audax and Peerless 1" soft-domes, had similar resonances between 9 and 16 kHz, which were partially damped by a small felt pad almost filling the space between the dome and the magnet polepiece. Since the soft-domes were much more lossy than the stiff B110 dustcap, the resonances were much broader and only 1 to 2 dB in magnitude ... but they were still there, and they were responsible for some of the fatiguing quality noticed by attentive listeners.

Not surprisingly, the problems were much worse in the phenolic, fiberglass, and hard paper domes used in the nastier speakers of the day. (Ah yes ... who can remember such paragons of excellence as the BIC Venturis? Cerwin-Vegas? JBL L100's? Once upon a time I actually *sold* these things!)

Returning to the present, the best midbass and tweeter drivers now sidestep the dustcap/dome problem in two ways: a vented polepiece assembly, used by the Scandinavian manufacturers Scan-Speak, Vifa, and Seas; or a bullet-like extension of the polepiece, which replaces the midbass dustcap entirely, used by the French manufacturers Audax and Focal.

The Scan-Speak D2905 series of tweeters are the most notable examples of tweeters with vented polepieces that load into a tiny transmission-line behind the magnet assembly. (The line progressively absorbs the backwave from the tweeter dome, improving the impulse response and power-handling of the tweeter.)

Magnetic Non-linearities

Most audiophiles are aware that loudspeaker drivers are inductive; after all, the voice coil is wound around a ferrous polepiece, and that's how you make an iron-core inductor. Not as many audiophiles know about the myriad of problems this creates.

If the inductance were constant, like an air-core inductor, there would be no problem; just adjust the crossover design to allow for it (using a simple R-C network) and off you go. Unfortunately, this is an *iron*-core inductor, and much worse, the inductance varies with the position of the voice coil.

The varying inductance has profound consequences, since the inductance is actually a important factor in determining the upper rolloff frequency of the driver, as well as its acoustic delay (relative to the tweeter). Vary this inductance, and the rolloff frequency and acoustic delay vary too. When does this happen? Whenever the driver moves a significant proportion of the linear region of voice-coil travel, which is a shorter distance than you'd think. In the excellent 8" Vifa P21W0-12-08, this linear region is only 8 mm (plus/minus 4 mm either way). A more typical figure for linear travel would be 6 mm for most 8" drivers, and 1 to 3 mm for most midranges.

Play some deep bass, and the effects of inductance modulation begin to show, creating IM and FM sidebands over the entire frequency spectrum. This is a genuine problem for 2-way systems and 3-way systems using a low midrange crossover; it means that any time you can actually see the drivers move, there are quite significant amounts of IM and FM distortion. What does this sound like? You can expect a "grayish" coloration and a blurriness that will change depending on the type of music you play.

Are there solutions? Yes. The drivers from Scan-Speak (SD System), Dynaudio (DTL-System), and the new Seas Excel series plate the polepiece with copper to short out eddy currents induced within the magnet structure by the voice coil. The specification that gives this away is the lower-than-usual voice coil inductance.

The 8" Scan-Speak 21W/8554, probably one of the best 8" drivers in the world, has an inductance of 0.1mH, which is far lower than the 8" Vifa P21W0-20-08, which has in inductance of 0.9mH. Both are excellent drivers; the Scan-Speak, though, is almost certainly going to have more transparent sound when asked to play bass and midrange at the same time.

The inductance figure has another hidden consequence; remember, the upper rolloff frequency of the driver is the combined function of the mechanical rolloff and self-inductance of the voice coil. If you calculate the electrical rolloff frequency by using the VC inductance and the DC resistance, a few drivers have an electrical rolloff well above the measured acoustical rolloff. This is *good*; it means that the interaction between the two rolloff mechanisms is going to be small.

Most drivers, though, have an electrical rolloff well *below* the measured acoustical rolloff. How is this possible? The mechanical system actually has a broad peak which is masked by the self-inductance of the voice coil. This is *not* good; any change in either the mechanical system or the electrical system is going to strongly modulate the frequency and transient response.

This, by the way, is the same kind of problem found in the old moving-magnet phono cartridges. Most moving-magnets (typically Shure and Stanton) were mechanically peaked, then rolled-off electrically by the combination of cable capacitance and cartridge inductance. By contrast, moving-coils have less than one-tenth the inductance, a much flatter and wideband mechanical system, and are much less susceptible to cable coloration.

The same applies to loudspeakers; it is always preferable to have a flat mechanical system and avoid compensating with electrical equalization; the trick is to know when this has been done within the driver itself by generous amounts of self-inductance.

Selecting A Driver

I use a method that's so crude it might sound dumb; I put the driver on large, IEC-sized baffle (135 cm by 85 cm) and listen to it! No crossover, no enclosure, and if it's a tweeter, not loud at all. I listen to pink noise (to assess the severity of the peaks that may appear in the sine-wave and FFT waterfall measurements) and music (to get a sense of the driver's musicality and resolution).

This does take an educated ear, though, since you have to listen around the peaks that the crossover might notch out, and not hold the restricted bandwidth against it. However, this listening process tells you a lot about how complex the crossover has to be, particularly if you remember that the crossover can never totally remove a resonance ... it can just make it a lot more tolerable.

I also carefully assess the results of the MLSSA PC-based measurement system (using the same IEC baffle), looking at the:

1) *Impulse Response*. How fast does the driver settle to zero? Is there chaotic hash in the decay region or is it a single, smooth resonance? Are there two or more resonances?

2) *Group Delay vs. Frequency Response*. How ragged is the frequency range above the first breakup? Can it be corrected in the crossover?

3) *Waterfall Cumulative Decay Spectrum*. Can you accept the resonances that can't be fixed in the crossover? If crossover correction is required, how complex is it going to be?

4) *Flatness of Frequency Response*. Take a good look at the Fletcher-Munson curves; these show where the ear is most sensitive to small deviations in response. The most critical region is between 1 and 5kHz; any peaking in this region, even as small as 1/2dB, is audible as an unpleasant "canned" quality. By contrast, small valleys are much less audible, so long as they are not caused by reflections or resonances.

Paying attention to these small details is the difference between the cheerful DIY builder with a table saw and the serious, dedicated craftsman(or craftswoman) who loves the art. As in traditional crafts, a deep knowledge of technical means is combined with a sense of beauty and purpose.

Direct-Radiator Drivers

It helps when you start listening and comparing to have a good grasp on the basic characteristics of the driver, so you can determine if it is a good example of its type. By listening carefully to the driver in an open baffle with no crossover, and examining all of the important specifications, you can find out just how well the designers solved the problems of making that particular type of driver.

Soft-Dome Tweeters

These tweeters, using silk domes with damping compounds, came into common use in the early Seventies with the introduction of the Peerless 1" soft-dome (remember the tweeters of the original Polk speakers?), followed by the superior Audax 1" tweeter, which found its way into many British and American designs during the 1970’s and early 1980’s.

These designs fell into disfavor with the introduction of modern titanium and aluminum domes, which swept the Audax-class soft-dome drivers off the audiophile market.

Over the last ten years, the soft-domes have made a surprising comeback with the gradual improvement to the Scan-Speak D2905 series of 1" tweeters, which compete on even terms with any metal-dome around. These tweeters combine vented pole-pieces with sophisticated transmission-line back-loading, new dome profiles, and new coating materials. As a result, they have the sonic resolution and detail of the best metal domes without the characteristic 22 to 27 kHz metal resonance.

Strengths are: Intrinsic self-damping and potential for extremely flat response and first-class impulse response. Potential for natural, open sound without intrusive and fatiguing resonances, a valuable quality when listening to many digital recordings.

Weaknesses are: The first-generation of soft-dome tweeters had a dull sound with a hard-to-pin-down fatiguing quality, as well as quite limited power-handling which required a high-slope 18 dB/Octave crossover. Modern soft-domes have solved these problems, with the best examples comparing to the best of all other technologies, including electrostatic and ribbon tweeters.

Best Examples are: The Scan-Speak D2905 family of 1" dome tweeters. I’ve used the Scan-Speak D2905 in the Ariel loudspeaker and am very pleased with the sound.

Soft-Dome Midranges

These are enlarged (2 to 3 inch) versions of soft-dome tweeters, using similar construction techniques with a half-roll surround acting as the combined surround and spider. Unfortunately, what works for a tweeter doesn’t work so well when scaled up for midrange use. In a tweeter, excursion requirements are modest (0.5 mm is plenty), but the requirements for the 3rd derivative of excursion (jerk) are severe, since the tweeter handles the very top of the spectrum, and is occasionally exposed to ultrasonic clicks from amplifier clipping, phono cartridge mistracking, or high-frequency noise and distortion from digital converters.

By contrast, the midrange (or midbass) driver experiences much greater demands for excursion and acceleration for two reasons: if you halve the frequency, you need four times as much excursion, *and* the musical spectrum carries most of its power in the lower midband. Both factors combine to make the midrange driver a device that must handle *much* more power than a tweeter. This imposes harsh demands on the rigidity of the diaphragm, and it exposes the simple suspension to rocking modes.

The reason conventional cones have a separate surround and an inner spider is to constrain the cone travel to a back-and-forth piston motion. Only very expensive mid domes intended for professional studio monitors (like the ATC) use a separate spider; as a result, most consumer-grade domes have serious problems with side-to-side rocking and other spurious motions. In addition, the doped-silk diaphragm is easily deformed by the high acceleration loads in the power band of the midrange. You don’t see bass drivers made out of doped silk, after all.

As a result of these problems, soft-dome midranges measure well, but sound a lot worse than conventional steady-state measurements would indicate. Even if you stick to measurements and discount all of the foregoing, they are limited bandwidth drivers, requiring a 12dB/octave crossover no lower than 500Hz (800Hz is better) thanks to a linear excursion of no more than 2mm. You’d expect a big tweeter to do well at high frequencies, but all of the soft-domes I’ve seen start to roll off at 4 to 5kHz, which is no better than good modern midbass drivers.

Of course, there are exceptions to what I’ve mentioned above. For example, there are cone-dome hybrids, such as the 5" Scan-Speak 13M/8636 and 13M/8640, and the 5" Dynaudio 15W-75. These new drivers are actually designed as high-quality miniature cone drivers, *not* as midrange domes. The only thing they have in common with the traditional soft dome is a large dustcap, which does indeed act as a dome radiator at the higher frequencies.

These new cone-domes have much more excursion, much lower distortion, and a much wider frequency response than the older soft-dome midranges. The cone-dome drivers are capable of realistic and transparent sound. They are described in more detail in the other sections, since they use Kevlar, paper, and polypropylene cone materials.

Another "special case" is the English professional-grade ATC 3" dome with an integral short horn. This driver uses a dual spider to eliminate the rocking problem that plagues most soft-domes, reducing the IM distortion very significantly. Ron Nelson (of Nelson-Reed) recommended this driver as one of the very best midranges around, and I take his recommendation seriously. This is a very expensive driver (around US$300/each). They *also* need to be hand-selected so the resonant frequencies of the left and right channels match.

Strengths are: None. Metal-dome midranges have some potential, but they require sharp crossovers on both ends with an additional sharp notch filter at high frequencies to remove the first (and worst) HF breakup mode. *Note:* This does not apply to the cone-dome hybrids or the prosound ATC driver.

Weaknesses are: High distortion, fatiguing sound, high crossover frequency, limited bandwidth, limited power-handling, and misleading frequency response measurements. It takes a detailed swept IM distortion measurement and laser holography to get the full story on these drivers. *Note:* As before, this does not apply to the cone-dome hybrids or the prosound ATC driver.

Best Examples are: ATC 3" professional-series - a totally different animal than the usual soft-domes. About 4 times as expensive, though (so what did you expect?). The Scan-Speak 8640 and 8636 are also excellent wideband midrange drivers.

Metal-Dome Tweeters

Advances in German metallurgy (at Elac and MB) resulted in thin profile titanium and aluminum domes in the mid-Eighties, with drivers from several vendors in Germany, Norway, and France now available. This type of driver can offer very transparent sound, rivaling the best electrostatics if correctly designed.

The downside is the lack of self-damping, with aluminum coming a little ahead of titanium in being better behaved in the ultrasonic region. At the present, all metal-dome drivers have significant ultrasonic peaks, ranging in magnitude from 3 dB (excellent) to 12 dB (not so good).

There's controversy about the significance of this ultrasonic peak, since the engineers of Philips and Sony have gone to great lengths to ensure that none of our wonderful new "Perfect Sound Forever" recordings ever have any musical information above 20kHz. Not withstanding limitations of the signal source, power amplifiers (and CD players) can generate spurious ultrasonic signals, especially at or beyond clipping. These ultrasonic signals can excite the metal-dome resonance, causing IM distortion to fold down into the audible region.

Strengths are: Uniform piston action right up to the HF resonance, providing sound of very high resolution, transparency, and immediacy if correctly designed. Dispersion is typically excellent, since the metal domes have flatter profiles than soft-domes.

Weaknesses are: Potential for (dare I say it) "metallic" coloration caused by the HF peak intermodulating with the inband sound. Some early designs have restricted power-handling. If overloaded, breakup distortion can be extremely harsh.

Best Examples are: Vifa D25AG-35-06 1" aluminum dome, which is even better with the plastic phase disk removed. This dome has a vented pole piece, so power handling is quite good, and the ultrasonic peak is only about 3 dB even with the phase disk clipped off (recommended). The Focal tweeters are reputed to be even better.

Ribbon Tweeters

The best-known true ribbon tweeter is the rare Kelly Ribbon of the Fifties, but other types appear every now and then. These are the only dynamic tweeters with the low mass, uniform drive, and low distortion of electrostatics. True ribbon tweeters are in a category of their own, since the of the design compromises of conventional drivers don't apply. Of course, that means they have a *whole new* set of problems! No free lunches in audio!

The biggest drawback of true ribbons is the one-turn "voice coil," freely suspended in the side-by-side magnetic gap.

This means the impedance and efficiency approach zero, unless a transformer is used. Even with a matching transformer, the efficiency of the ribbon was still pretty low, which is why Kelly added a short horn to their tweeter. Unfortunately, the short horn compromised some of the best traits of the ribbon, which are accurate pulse response and freedom from resonance.

By combining rare-earth magnets with an integral transformer, Raven has raised the efficiency of their ribbons to an astonishing 95dB/metre. This is *ten times* higher than traditional ribbons, and without horn loading! The MLSSA waterfalls look impressive too, but that's to be expected with ribbons, along with low distortion. (Raven claims less than 1% distortion at 105dB continuous output, a very good figure.)

The only drawback I can see for the Ravens is the requirement for a high-slope crossover. This is a potentially serious issue, since a 4th-order (acoustic) slope is already on the threshold of audibility, with a 360-degree phase rotation at the crossover frequency. The most direct way around this is raising the crossover frequency, and selecting a very wideband midbass driver.

Paper Cone Midbass & Full-Range

This class of drivers go right back to the original Rice & Kellogg moving-coil patent of 1925. Paper-cone drivers range in quality from terrible to wonderful; from a ten-cent speaker glued to a computer motherboard to the superb Scan-Speak 5" cone/dome midrange, the classic horn-loaded Lowthers, and the 12 and 15-inch Tannoy Dual Concentrics.

This oldest of cone materials is actually a composite structure, and changes properties significantly when treated with an appropriate material (the makeup of the additive is invariably a trade secret of the manufacturer). The cone treatment is quite important, since paper undergoes significant alterations with changes in humidity if left untreated; the additive stabilizes the material, improves the self-damping, and extends the HF response of the cone.

Strengths are: Good-to-excellent self-damping, potentially excellent resolution and detail, very flat response potential, and a gradual onset of cone breakup. It can be used with low slope linear-phase crossovers without much trouble. Paper is a material that sounds better than it measures … this is an genuine asset, not a disadvantage.

Weaknesses are: Not as rigid as the Kevlars, carbon fibers, and metals, so it lacks the last measure of electrostatic-like inner detail. Doesn't go as loud as the materials above, but the onset of breakup is much more gentle and progressive. Paper-cone drivers may require modest shelving equalization in the crossover for the best results.

Paper is not as consistent as synthetics, so pair-matching isn't quite as exact, which may affect imaging, depending on the precision and quality of manufacture. Properties may slowly change over time, depending on the composition of the cone.

Best Examples are:

Scan-Speak 8640 5" cone/dome midrange, with linear response from 300Hz to 13kHz, very low distortion, excellent pulse response, and excellent inner detail.

Audax PR170M0 6.5" high-efficiency midrange. (100 dB at 1 meter!)

Diatone PM610A 6.5" (Anniversary Edition) from Mitsubishi in Japan. These are very wideband drivers, covering 70 Hz to 20kHz in a conventional enclosure.

Various Lowther models. These require horn-loading for correct operation, and to prevent over-excursion damage. If correctly horn-loaded, they cover a wide range from 50Hz to 18kHz.

Bextrene Midbass

This is an acetate plastic derived from wood pulp, and is typically damped by a layer of doping material on the front of the cone to control the strong first resonance it displays around 1.5 kHz. It was originally developed by the BBC in 1967 to replace paper with a more consistent and predictable material for monitoring purposes. It came into widespread use in the early Seventies, with the typical audiophile speaker using a 8" KEF or Audax Bextrene midbass driver with an Audax 1" soft-dome tweeter.

The BBC-derived designs always employed notch-filter equalization to flatten the Bextrene driver in the midband; the most famous (or infamous, depending on whether you were the listener or the designer) driver was the KEF B110 used in the BBC LS 3/5a minimonitor. Not everyone knows that this speaker, which is legendary for its sweet midrange, employs a deep notch filter with 6dB of attenuation at 1.5kHz to correct the B110.

Over time, Bextrene has been replaced by BBC-developed polypropylene, which gives much flatter response, does not require a layer of doping material, and provides a 3-4 dB increase in efficiency due to the decrease in cone mass. Bextrene is now considered an obsolete material by nearly all speaker designers.

Strengths are: Consistent batch-to-batch, excellent potential imaging (by mid-Seventies standards). Inner resolution higher than many paper cones.

Weaknesses are: Very low efficiency (82-84 dB at 1 meter), requires a strong notch filter in the midband, a "quacky" coloration by modern standards, sudden, unpleasant onset of breakup at not-so-high levels, and numerous resonances at the top of the working band.

Best Examples are: None. Modern designers are not willing to tolerate the low efficiency and the complex notching and shelving equalization required to make these drivers acceptable. Although some traditionalists revere the KEF B110 used in the Rogers LS 3/5a, the uneven response of this driver requires the LS 3/5a crossover to be very complex. Having worked with the B110 for many years, I feel the modern Vifa P13WH-00-08 is superior in every way.

Polypropylene Midbass

This material was developed and patented by the BBC in 1978 (my dates may be off) as a replacement for Bextrene. Since it is intrinsically self-damping, a correctly designed polypropylene driver is capable of flat response over its working range without little or no equalization. In addition, they typically attain efficiencies of 87 to 90 dB at 1 meter, which is a major improvement over Bextrene.

This material has become nearly universal, since it requires a minimum of hand treatment to assemble a loudspeaker - the only difficult problem was finding the cyanoacrylate adhesives that would stick to a slick material like polypropylene. That problem was solved in the beginning of the Eighties.

As with paper, this cone material is used in speakers ranging in quality from mass-fi rack-stereo systems to the first-rank ProAc Response series. The cone profile, termination at the edge of the cone, and additional materials added to the polypropylene mix strongly determine the ultimate quality of this type of driver.

Strengths are: Very flat response if correctly designed, very low coloration, good impulse response, gradual onset of cone breakup, good efficiency, and a crossover that can be as simple as one capacitor for the tweeter. The best examples can be as transparent as the best paper-cone drivers, which is a very high standard.

Weaknesses are: Not quite up to the standard of transparency set by the rigid-cone class of drivers and the planar electrostatics. Many poly midbass units do not mate well with the popular metal-dome tweeters, with differences in resolution that can be obvious to the skilled listener. Not a good choice for woofers 8 inches or larger unless the polypropylene is reinforced with another, more rigid, material. Woofers 10" or larger are better served by stiff paper or carbon fiber.

Best Examples are: The Scan-Speak 18W/8543 7" midbass, as used in the ProAc Response Threes, is probably the finest polypropylene driver in the world.

Another closely-ranked contender is the Dynaudio 17W-75 Ext. 7" midbass, as used in the Hales System Two Signature.

The Vifa P13WH-00-08 5.5" midbass/midrange unit is a superb performer, well suited for midrange or minimonitor use. It is unique in having a textbook-flat midrange combined with a completely smooth Bessel 2nd-order rolloff. I use these drivers in the Ariels, and I'm very pleased with the results. The Vifa P13WH does not have the typical "poly" sound, sounding much more like a top-rank paper-cone than other polypropylene drivers.

Rigid Midbass

Aluminum and Magnesium Drivers

The first rigid drivers to find limited use in high-fidelity applications were the small Jordan Watts 2" *aluminum-cone* units. The hand manufacture, high price, and low efficiency limited the market for these drivers, and very few appeared in the United States (I have heard of them by reputation but have never auditioned them personally).

There is a new generation of British and German 2-way minimonitors that use proprietary 5" to 6.5" aluminum-cone midbass drivers. These drivers typically have very low efficiency (82-84 dB/metre) and almost certainly have a high-Q peak at the top of the working range. Since these drivers are not being sold on the open market, I have not seen any detailed information on them.

The new Seas Excel series uses magnesium cones with an intriguing solid-copper "bullet" replacing the usual dust-cap. They certainly look beautiful, and have a quite usable efficiency around 87dB/metre, unlike the older aluminum-cone units above. The preliminary data I've seen, though, shows a whacking great 16dB peak at 4.9kHz, so you better be a pretty good crossover designer!

Expanded-Foam Drivers

The next generation were the *expanded-foam* bass units, with the KEF B139 being the most famous example. This class of driver offered piston-band operation through the midbass, but suffered from very low efficiency, limited power-handling, and severe high-Q resonances in the midband.

(It was not generally known that the B139 had a 12dB peak at 1100Hz with a very high Q. Many reviewers blamed the midrange for problems that were actually caused by the lack of a notch filter for the B139.)

They were quite popular in 3 and 4-way transmission-line systems in Britain (IMF) and the United States (Audionics) in the Seventies, which is where I enter the story ...

I remember working with the KEF B139, B110, Richard Allen 7" and front and rear KEF T27's in my very first commercial design, the Audionics TLM-200 4-way system. My baptism into the mysterious art of speaker design went as follows:

Charlie, my boss: *"Hey Lynn! You remember what Laurie Fincham was talking about when we visited KEF in England? All that slide-rule stuff about driver impedance correction and frequency response target functions?"*

Me: *(warily) "Well, I didn't write it down, but I remember a bit of it."*

Charlie: *"Great! You can do the crossover for THIS!"* pointing at a gigantic six-foot-tall loudspeaker with the 4 aforementioned drivers.

The original designer had high-tailed it to Seattle and disappeared without a trace, leaving behind the two monstrous prototypes, which at least were finished in an attractive walnut-veneer cabinets. No crossover had even been started, and even I, a rank novice, knew that crossover design was the single hardest part of making a new loudspeaker.

Six months later, with a 57-component crossover sitting on the floor, the TLM-200 was done. How did it sound? For 1973, not too bad. I think we sold maybe 10 of them. Fortunately, the other models I designed for Audionics sold in the mid-hundreds ... and I wasn’t stuck with the marketing department choosing the drivers.

Carbon-Fiber Drivers

The next generation or rigid-cone drivers were the Japanese *carbon-fiber* units, which made their first appearance in the pro studio monitor (prosound) 12" TAD units with very high efficiencies and very high prices (around $300 each in 1980). Carbon fiber prices have now dropped, and Vifa, Audax, and Scan-Speak make good examples of this type of driver. The Japanese make lots more of them, having pioneered the technology, but they are very difficult to obtain if you are a non-Japanese small-run specialist manufacturer.

These drivers have true piston action, outstanding bass and midbass response (the best I have ever heard), but also have a characteristic double-peak region at the top of the working range. Unfortunately, these peaks are grossly audible in most carbon-fiber drivers, and worse, cannot be removed by a notch filter tuned between the two peaks; it requires *two* notch filters to control the peaks, or a low crossover with a very sharp rolloff (24dB/octave) to remove them from audibility.

Although I very much dislike drivers that require filters as complex as this (after doing the TLM-200, I vowed never again to design a 57-component crossover), I must admit that carbon fiber woofers are the only direct radiators where I've actually felt tactile bass.

The Scan-Speak 18W/8545 looks pretty interesting; although it has the classic double-peak signature of a carbon-fiber driver, they look quite well-damped, and the breakup region above these two peaks looks pretty smooth. It may even be possible to use the Scan-Speak 18W/8545 with a simple 2nd-order filter.

Kevlar Drivers

Kevlar drivers made their appearance in the mid-Eighties with the French Focal and German Eton lines, with the Eton having superior damping due to the higher-loss Nomex honeycomb structure separating the front and rear Kevlar layers.

At the time of this writing, the 7 and 8-inch Scan-Speaks have the best frequency response and the lowest IM distortion of any Kevlar driver. Another desirable property of this family of drivers is a well-behaved rolloff region *above* the characteristic Kevlar peak. All of the other Kevlar drivers (that I have measured and listened to) have chaotic breakup regions; by contrast, the Scan-Speaks are the only ones that appear well-controlled in this region. This is certain to provide a significant improvement in smoothness and transparency.

Composite Drivers

Audax has an unusual *composite* technology, called HD-A. This is an acrylic gel containing a controlled mix of grain-aligned carbon-fiber and Kevlar fibers. The response over the main frequency range is impressively flat, with only a moderate peak at the top of the range.

Another intriguing series of composite drivers are the Focal Polyglass paper-fiberglass cone drivers, with the 6V415 midbass looking most interesting, with very flat response and quite good excursion capability. For the East Coast triode fans who are into the no-crossover full-range driver thing, a stack of four 4V211's might do the trick, with a response flat from 60Hz to 12-14kHz.

Strengths & Weaknesses of Rigid Drivers

Strengths are: Best available transparency, imaging, and depth presentation of any type, equaling or exceeding electrostats if carefully designed. High efficiency, high peak levels, and very low IM distortion in the best examples. This class of drivers is considered at the state of the art by many designers, and this field is expected to advance quite rapidly as material technology advances.

Weaknesses are: Older designs have severe peaking at the top of the working band, and most have a uncontrolled chaotic breakup region above this high-frequency peak. This would cause fatiguing sound over the long run and a compression of depth perspective and "air."

Loudspeaker systems that does not use correctly designed notch filters with a metal, Kevlar or carbon-fiber drivers can be considered faulty, since the narrow HF peak does not lend itself to correction with a standard low-pass filter. The sound of this peak will be obvious to any listener familiar with the sound of an unequalized Kevlar or carbon-fiber driver. (Tap the cone to hear this.) The new 7-inch Scan-Speak 8545 and 8546 may be the first of a new generation of moderate-peak drivers that *won't* need the usual notch filter.

Although these types play quite loudly, the onset of breakup can be harsh and unpleasant, akin to clipping in a amplifier. Some Kevlar and carbon-fiber drivers require an extremely long break-in period (>100 hours) to soften the fibers in the cone and the spider.

Best Examples are: The Scan-Speak family of 5", 7", and 8.5" Kevlar and the new 7" carbon-fiber/paper drivers. These are the only rigid drivers that have well-damped peaks and a reasonably well-behaved rolloff region above the main high-frequency peak. In addition, the Scan-Speak drivers also have vented pole-pieces that are copper-coated, reducing inductive types of IM distortion by tenfold or more.

The new Seas Excel series with magnesium cones and solid-copper phase plug look interesting if the designer is willing to meet the challenge of designing a very deep and accurate notch filter to correct the first resonance of the magnesium cone.

Some Closing Thoughts On Speaker Design

Don't be led astray by marketing literature … *all*, repeat *all*, drivers have a sonic signature, which can only be controlled, not eliminated, by equalization in the crossover network. Even though crossover equalization can straighten out the driver in the frequency and time domains, the IM distortion still undergoes a shift in character when the diaphragm or suspension goes into a resonant mode. All physical materials have resonant modes, so if the driver is constructed of physical materials, it'll have resonant modes!

Since we all have to work with imperfect materials, here's a set of design guidelines that can help us get from the world of abstract ideas and concepts to happy listening.

- Aim for a reasonably smooth frequency and impulse response, and take steps to eliminate any reflections from the front of the cabinet or from the interior surfaces. Reflections are much more audible, and much more unpleasant, than you'd imagine from looking at the little wiggles they make in the frequency response curves. On the inside of the cabinet, intelligent use of felt (85% wool) and Deflex pads can usefully damp reflections. On the outside, *never* put the driver in any kind of cavity, since even the best felt absorber has only a modest absorption capability. It's far better to mount the faceplate precisely flush with the front panel and radius the cabinet edges if at all possible.

- Reduce driver and cabinet resonances to levels where they aren't too obtrusive. Cabinets in particular must be rigid first, with damping a secondary priority. Interior crossbraces of 3/4" high-grade plywood that run the full interior width of the cabinet can make all the difference here.

- Avoid crossovers in the critical 300Hz to 3kHz region. The telephone company was correct in selecting this region as the most important part of the spectrum; this is the region that must be really spotless, with flat response and very low distortion. Even if the crossover is brilliantly executed, using the most modern computer tools and months of subjective balancing, it is still slightly audible. That is why it is better to keep the crossover out of this sensitive frequency range.

- A well-controlled and peak-free response in the rolloff region of the crossover is very important; if this can be done, you get a smooth phase and amplitude hand-off between the drivers, relaxed and sweet midrange, and greatly improved image quality.

- Last but not least, reduce the IM distortion across the spectrum, with greatest emphasis on the 500 to 5 kHz region. This means selecting drivers with well-designed magnetic systems and using midranges and tweeters that are generously sized for the intended frequency range (5" to 7" midrange and 1" tweeters).

Resources

Test and Measurement

DRA Labs MLSSA
The standard of the industry and the system I use myself. Not cheap, and requires a separate calibrated lab-grade condenser microphone and preamplifier from Aco Pacific.

CLIO from Audiomatica
The CLIO hardware/software system includes a calibrated microphone and is a professional-grade instrumentation system. This is an important feature, since calibration accuracy and data-file compatibility with separate crossover-optimizer programs is essential for serious design work.

Pen-Strobe from MLS Instruments
Very useful for analyzing what a speaker cone is really doing. Click the "Voicecoil Article" for an interesting description of using the strobe to detect rocking modes, standing waves in lead-out wires, tweeter dome resonances, direct-observation phase testing, etc. An essential tool if you plan to modify a driver.

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