AUDIO Perfectionist™ JOURNAL

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Speakers

PART 1

INTRODUCTION

by Richard Hardesty

*In previous **Journals** we've explained at length why loudspeakers can't reproduce a signal that your other components don't send them. The following is also true: a bad set of speakers can take a perfect signal and make bad sound.*

Are the speakers that reproduce the signal more important than the system that creates it? That question has no definitive answer because you can't listen to either separately. While it's unrealistic to expect that a great set of speakers will make a bad system sound better, it's just as foolish to think that you can get great sound without great speakers.

In this **Journal** we'll start from the beginning and explain how speakers work and then describe why most of them can't begin to accurately reproduce the signal that your audio components created.

Speakers are by far the most inaccurate and colored part of an audio system. Why should they be accurate, you might ask, if they sound good to me? Let me tell you.

Journal Explained

The **Audio Perfectionist Journal** takes a different path as we try to help you choose a satisfying home audio system. The **Journal** follows Richard Hardesty's high fidelity approach to audio reproduction. The high fidelity approach states that good audio components should accurately reproduce the recording, as opposed to those components which "artfully" create a pleasing sound, which might stimulate some listeners in a way that reminds them of some music, some of the time.

How does an accurate audio system differ from one that sounds good to some people when playing some types of music? First let me define high fidelity and then I'll tell you why I think it's the best way to choose components.

Why The High Fidelity Approach?

An audio system creates a reproduction of the original event. The result is a simulation. Real musicians aren't playing in your listening room—but it should sound like they are. If a certain coloration could be proven to improve all types of music it would be hard to argue against it. No such coloration has ever been found.

> *"…good audio components should accurately reproduce the recording…"*

Instead, some components are designed with inaccuracies complimentary to certain kinds of music and detrimental to others. Many audio components and most speakers fall into this category. My observations have led me to believe that this is not the best way to choose components or assemble systems.

If your car system can satisfyingly deliver the repetitive thump of rap bass it probably won't reproduce a cello worth a damn. If you listen to nothing but rap this coloration may be acceptable, even desirable. If you enjoy a wide variety of music, and I believe that most audio perfectionists do, then you'll want an accurate audio system that faithfully reproduces the recording—whatever it contains.

Let me Share my Wisdom Grasshopper

I have observed hundreds, perhaps thousands, of folks on the same quest. Their common goal is a satisfying musical experience in the home. I have found that more people will be happier longer with a high fidelity audio system that accurately reproduces the recording.

The engineers and technicians who make the recordings try to produce an accurate replica of the live performance they recorded. Their results vary but for the most part they do a pretty good job. If you assemble a system that accurately reproduces what they recorded you are likely to hear a pretty good representation of the original event.

If, on the other hand, you choose an amplifier or speaker system that adds a pleasing coloration you are likely to find that the enhancement works only for a certain type of recording and negatively impacts all others. Artificial ambience, for instance, may enhance the reality of classical music recorded in a large venue but it is unlikely to add realism to an intimate recording of a solo vocalist accompanied by a single guitar.

The Journal Format

Each **Audio Perfectionist Journal** concentrates on a certain section of the audio system. We start by explaining how a certain component works. We highlight the sections of that component that make the biggest sonic differences. We gather a selection of these specific components, which contain the various design choices, and describe what we heard when evaluating these components in direct comparison, in the same room using the same reference system and the same pairs of ears.

Journal 12 is all about speakers but time and space limitations mean that we'll only scratch the surface of this complex subject. Speaker function will be explained in terms you can understand and an informative interview, with one of the best speaker designers in this country, will be included. I'll tell you why most speakers sound like nothing you've ever heard before—and why that's bad—and I'll review the speakers that I chose to buy for myself. APJ

Bias or Astute Selection?

by Richard Hardesty

Are we biased towards certain components? I know I am. I purchase only the best things I can afford. I choose carefully and with good reason. I am certainly biased towards the things I have chosen and against the things I have rejected. These biases are reflected in my writing in the ***Journal****. How does that compare with advertising-supported publications?*

Freedom From Advertising versus Freedom From Bias

Publications that are supported by advertising cannot risk offending advertisers. They deny this fact but it's true and easily verified.

In a perfect world a free press would provide thoughtful criticism. In the real world, salaries of the writers of the not-so-free press are paid by advertisers. The costs of producing, printing and mailing a magazine are also paid by the advertisers. Readers often pay less than the cost of printing and mailing the publication because the subscriber list establishes advertising prices. Magazines are written for advertisers not readers.

> ***"In a perfect world a free press would provide thoughtful criticism."***

Everything "reviewed" in an advertising-supported publication will be praised and any criticism will be soft-pedaled or hidden between the lines. Who would buy advertising space in a publication that printed a negative report about their product?

Advertiser Bias, Price Bias

I read denials of bias in magazines every day. The editorial staff will claim complete separation from those who sell advertising. Is this true? Ask them to show you some of their product criticisms. Is everything they review good?

Do they simply ignore products that aren't any good or those that cost far more than their performance justifies? Or does every

component perform better than anything that costs less, as the magazines would have you believe?

Owning Your Own Components

Free components are another major factor influencing magazine reviews. This bias is consistently denied and you may question who is telling you the truth. I challenge you to write to any magazine editor and ask what audio components they have actually purchased for their own use. Discounts don't count. Editors can get a discount on anything, which levels the playing field. We're talking about free stuff. Did they buy their equipment or did they get it for free?

Do they really like the carriage trade products they rave about? Do they fork over their own cash to buy these products or do the manufacturers give them "long term loans"? Would a magazine reviewer dare to announce that certain components outperform the ones the advertisers have provided? What would he or she listen to if the manufacturers got angry and asked for their products back?

It's true that manufacturers seldom loan products with a specific agreement that they will receive positive ink. They don't have to—It's implicit.

Is the Journal Free From Bias?

We are certainly not free from bias. Would you listen to the opinions of someone who didn't have any?

Of course we're biased. We evaluate groups of the best products we can find and provide candid opinions about the sound we heard, even if it falls short. We buy the best examples with our own money if we can afford them. We tell you when something sounds better than what we can afford to purchase and we fear no retribution from the manufacturers of the components we use because we own them! Do I need to explain this difference any further? The **Journal** is written for you. There are no advertisers involved.

> ***"Would you listen to the opinions of someone who didn't have any?"***

Our biases represent our real, honest opinions based on actual product performance. Magazine writers can't even tell you what they really think. They work for advertisers not you.

In the **Audio Perfectionist Journal** you'll get real information about home audio systems.

Before we start the discussion of loudspeakers I'd like to bring you up to date on a couple of items that I've added to my reference system. Here is a review of the Solid Tech rack system and the story of the evolution of the Wadia 800-series CD player. APJ

Solid Tech Racks Review

by Richard Hardesty

Your audio components must be supported and displayed on something and that something may negatively impact the performance of those components. I've discovered a rack system that actually sounds better than most. It looks good, too, and it can be assembled in a wide variety of ways to suit almost any system configuration.

Mass and Isolation

A heavy object is hard to move. A massive equipment rack is a heavy object especially if it's loaded with high quality components. If large forces have a difficult time moving an object like a high-mass equipment rack it seems like smaller forces would have little effect at all. Based on these observations it would be easy to assume that a massive equipment rack would provide good isolation from the various vibrations that affect the sound of your components.

Contrary to intuition a massive rack, coupled to the floor with spikes, could actually introduce more vibration into your audio components and blur the sound they produce. How can this be? Let's forget about audio for a minute and look at the situation as purely a matter of physics.

High Mass and Energy Storage

Mass is related to the weight and density of the material being considered. Total mass will include the objects that are mechanically coupled to that material. A heavy-duty equipment rack loaded with high quality components will be a formidable object with a lot of total mass. Massive structures look cool but do they minimize vibration or simply change the frequency of that vibration?

> ***"…a massive rack, coupled to the floor with spikes, could actually introduce more vibration into your audio components…"***

If we ignore damping and methods that convert energy into heat we find that everything vibrates and everything will resonate at some frequency. Objects with higher mass will vibrate slower and resonate at lower frequencies. Lower mass objects will vibrate faster and resonate at higher frequencies. More massive objects can store more energy because there's more storage material. Objects that resonate at lower frequencies will store energy for longer periods of time because the period of a lower frequency is longer than the period of a higher frequency. These facts may lead us to the following conclusion, although I readily admit that there's more to it than I'm capable of explaining.

Higher mass objects can actually store and release more energy after a greater delay in time. This is not a desirable characteristic for an equipment rack. Since most audio components are built on a chassis that provides some isolation from vibration entering from the shelf below (at audible frequencies), the most desirable characteristics in an equipment rack might instead be described as structural rigidity combined with low mass and isolation of each supported component from the other components and from the rack as a whole.

Large Surfaces Exposed to Sonic Vibrations

An audio system usually includes loudspeakers that fill the room with energy in the form of sound pressure waves. Sound

pressure waves will have a greater effect on large exposed surfaces because there will be more area to intercept and vibrate in sympathy with the sound pressure variations. Big horizontal shelves will expose maximum surface area to the effects of these vibrations.

A stack of 3 to 7 of these shelves, closely coupled together, will tend to make the rack and all the components sitting on it vibrate as one large mass. Cones or spikes between these shelves, or between components and shelves, will assure that everything is coupled together both horizontally (if the spikes are attached) and vertically (whether the spikes are attached or not).

Based on these observations it appears that shelves should be eliminated and each component decoupled from the rack structure if possible. Components should be supported on the minimum possible contact area and horizontally decoupled (at least) or completely decoupled with a spring suspension system that isolates each support from the rack assembly.

The Solid Tech Solution

Björn Ohlssona, a crafty Swedish toolmaking engineer and audiophile, was not pleased by the available component racks and set his sights on a better solution. The result is the Solid Tech *Rack of Silence.* This equipment rack combines all of the sonically desirable characteristics into a practical device that supports all your audio and video components and looks good in your listening room.

The *Rack of Silence* has a skeletal structure with four extruded aluminum pillars and no high-mass components. There are no cones or spikes and no center pillar in the back to interfere with cabling. Vertical columns are available in three heights and the racks can be configured in a variety of ways.

With the *Rack of Silence* the usual shelves are replaced by ridged, low-mass cross members with minimal surface area. The components are supported on steel balls that decouple them from horizontal movement. The cross members can be suspended with springs, which act as low-pass filters and provide additional isolation, separating the components from each other and the rack structure. The attractive racks are fully adjustable and can be customized to suit your exact component requirements.

Do They Sound Better?

I have compared the *Rack of Silence* to several massive component racks and believe that the Solid Tech racks do provide better sound. The differences you'll hear vary from component to component but can be quite substantial. I have heard some improvement in virtually all cases.

The *Rack of Silence* is particularly effective with devices which may be adversely impacted by vibration such as turntables, disc players and electronic components that utilize vacuum tubes, but they also seem to provide audible benefits with

solid-state amplifiers. The Solid Tech *Rack of Silence* tends to make everything sound a little less mechanical and a little more musical. I bought five of them. My sensitive components are supported by suspended cross members.

Solid Tech components are imported into the USA by Audiophile Systems, Ltd. For more information visit their web site. http://www.aslgroup.com APJ

REVIEW

Wadia 861se INTEGRATED CD PLAYER

by Richard Hardesty

I have used a Wadia 800-series CD player as a digital reference source for many years. I started with an 860, which was upgraded to an 861. My player is now an 861se. Since I have lived with each one I thought readers would be interested in my experiences and Wadia's progression in performance.

The CD As a Source

The "Red Book" compact disc is inadequate as a high quality source for audio. This has become glaringly apparent in recent years with the introduction of high-resolution digital formats and the final realization by many music lovers that an analog turntable, playing "obsolete" vinyl records, simply sounds better.

In spite of this I have a wall covered with a large selection of CDs. In many cases the music I want to hear is available in no other format. CDs are the most popular almost-high fidelity source available and will be for some time. And they play in my car.

I have published a number of eulogies for the compact disc with "I told you so" glee because, well, I told you so. I disliked the format when it first arrived more than twenty years ago because I recognized it as a giant step backward in fidelity. I tried to get people to buy records and turntables instead of the convenient new compact discs and players and suffered financially because of it. Compact discs were definitely more user-friendly and the ads told us that they offered "perfect sound forever." Most people fell for it.

I resigned myself to the CD format, and both the players and the recordings improved immensely over the years. I continued to use records and SACD discs for evaluation and demonstration because the sound was so much better but I often listened to CDs for pleasure because much of the music I wanted to enjoy was available only on compact discs. Then something happened that changed my formerly ridged position.

"This player changes everything!"

Wadia introduced a new player, the 861*se*. It might slip by as just another upgrade to a familiar product because it still has

the old number with only an *se* designation added, but believe me, this is more than a Special Edition of an old standby. This player changes everything!

Wadia Changes the Picture

I have used a Wadia CD player for many years and compared it to every contender that came along, including separate transport/DAC combinations. It started as an 860 and was upgraded to an 861. When Wadia introduced the 861*se*, a Special Edition version of the 861 with an entirely new transport, they were eager for me to hear it. I agreed to let them upgrade my player yet again, well aware of the risk that I might not like it better.

I felt that my experience with the three versions of this machine would make interesting reading and was worth the gamble on the sonic expertise of the engineers at Wadia. I wasn't prepared for what I got back—it's hard to imagine that a mechanical change could make such a difference! The 861*se* makes listening to compact discs a generally satisfying musical experience at last.

My History with the Wadia CD Players

I got my Wadia 860 in the 1990s. It was the best available CD player then and it has remained the best performer ever since due to a series of improvements described below. I was momentarily seduced by some performance aspects of other offerings but always returned to the Wadia because it proved to be the most enjoyable player for compact discs.

There are some common traits shared by all three models described here so we'll start with those. All three versions have monolithic aluminum construction, dual isolated power transformers, the DigiMaster™ filtering system, and the ClockLink™ jitter elimination system utilizing full-clamping disc transports slaved to master clocks placed near the DAC chips.

Monolithic Chassis

The Wadia chassis is robust to say the least. The fifty-pound unit feels like a solid block of steel but is made entirely of machined aluminum. This beautiful enclosure provides a highly effective shield against radio frequency interference and vibrations that might adversely affect the delicate electrical and mechanical components. There is a separate isolated chamber for the transformers, which provides shielding for stray magnetic fields and mechanical damping for transformer vibration while allowing a low-impedance path for instantaneous power delivery from transformers to power supplies within the main chassis.

Dual Transformers

The Wadia players have two transformers. One transformer powers the transport and servo controller and the other one drives the digital-to-analog circuitry. This prevents digital noise from contaminating the analog circuits. The transformers “float” in a machined aluminum isolation chamber and are clamped into position by neoprene damping material. They are not otherwise mechanically attached to the chassis. You can’t hear them in the room or through the circuitry.

DigiMaster™ Upsampling Filter

The patented DigiMaster™ software runs on a powerful DSP computer and offers 24-bit resolution. It upsamples to 1.4112MHz (32x) and provides a digital reconstruction filter with superior time and phase performance. Long before “upsampling” became a buzzword Wadia was doing it. The Wadia machines always delivered superior imaging with greater depth information and a more musical sound compared to ordinary CD players because they perform better in the time domain.

ClockLink™ Jitter Elimination System

Most CD players have a single clock placed near the transport and the clock signal is transmitted to the DAC chips, which may induce jitter. Jitter is a well-known source of distortion because the digital code must provide the amplitude of the analog signal at exactly the right time or disaster ensues. Wadia’s ClockLink™ system locates the master clock at the DAC chips and slaves the transport to it, virtually eliminating the problem of transmission-induced jitter. Full clamping transport mechanisms reduce mechanically induced jitter.

Transport Superiority

The 800-series has always used a transport mechanism that clamps the entire disc to the platter for vastly superior mechan-

ical integrity. This makes a bigger difference than I thought and my eyes were opened (wide) by the audible improvement still possible through mechanical means.

The dCS Verdi, the ARC CD3 MkII and the new *se* version of the Wadia 861 have convinced me of the importance of the

transport mechanism. No, you can’t use just any old CD player as a transport and get great sound. The Wadia transport is the most impressive in terms of mechanical integrity and sounds the best in my opinion.

The Wadia players are not simply modified mass-market machines. They are thoroughly engineered products where every detail has been considered. Let me offer my impressions of the three stages of development that I’ve observed with the Wadia 800-series CD players. I’ve lived with each one for an extended period now and my opinions are well considered.

The 860

The Wadia 860 was clearly a superior player in its day. It delivered a more three-dimensional image and a consistently more musical presentation than other CD players. Shane and I used it as a reference for comparison as we auditioned dozens of other players. Then along came the Ayre D-1x. The Ayre player offered a more detailed presentation and provided the first real challenge to the superiority of the Wadia 860. The Ayre had the new Burr-Brown 1704 DACs and a zero feedback current-to-voltage converter. This challenge was answered by the Wadia 861.

The 861

The Wadia 861 retained the musicality of the 860 and added the detail and dynamics of the finest players, some of which had surpassed the 860 in those areas. The 861 had new Burr-Brown 1704 DACs, a revised DigiMaster™ filtering system (version 2.4) and Wadia's patented Swift Current™ current-to-voltage converter with no negative feedback. This player stood alone at the top of the heap and still would—but the Special Edition was introduced anyway.

The 861se

The Wadia 861*se* sets a whole new standard! The most significant change offered by the Special Edition is an entirely new transport mechanism. This transport has improved engineering precision on all surfaces including the clamping device and platter, which is now an aluminum and brass hybrid stained green to reduce undesirable laser reflections.

The platter is supported by a composite bridge composed of 20mm-thick machined aluminum and 5mm-thick carbon tool steel. This new transport allows the 861*se* to deliver blacker backgrounds, even better imaging than before, and resolution of detail that I previously didn't believe was on the recording.

Conclusion

If you have a collection of CDs—and who doesn't—you'll be glad to learn that there actually is satisfying music hidden on them. If you have a Wadia 860 or 861, an upgrade to the Special Edition version is a no-brainer. If you want the best available CD player, buy a Wadia 861*se*. APJ

Now let's examine the perplexing subject of loudspeakers. We'll start by explaining how they work in the simplest terms possible.

How Drivers Work

by Richard Hardesty

The most basic elements of a loudspeaker are the drivers and the crossover network that divides the signal into frequency ranges. We'll discuss those parts first, then we'll examine how they are assembled into complete speaker systems.

What is a Driver?

A drive element, often simply called a driver, is a mechanical component of a loudspeaker that actually makes sound. This speaker component converts electrical energy from the audio signal into mechanical energy that you can hear.

A driver has a moving diaphragm that stimulates the medium in which we are immersed, usually air, which in turn stimulates our eardrums allowing us to hear the sound the driver makes.

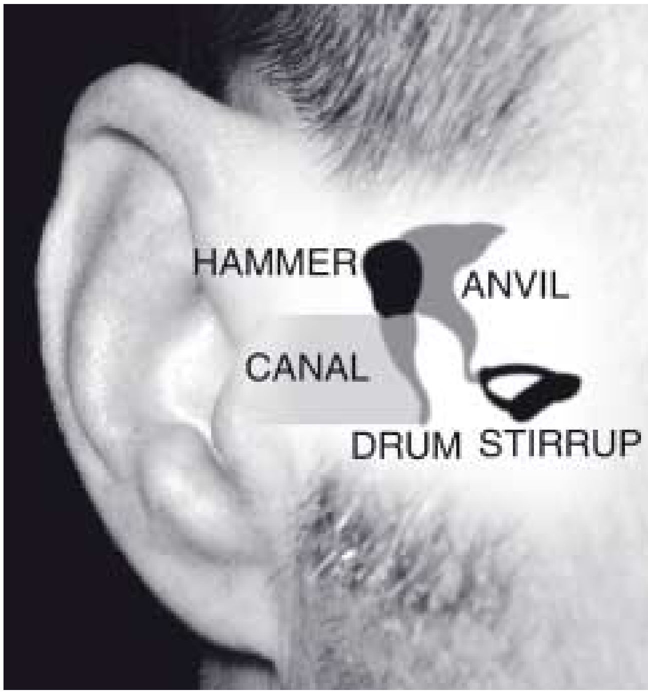
Development of a truly full-range driver has proven to be illusive and no drive element, despite claims to the contrary, can actually provide full-range response. Most speakers, even planar designs, employ two or more drivers in order to reproduce all frequencies within the range of human hearing.

A driver specifically designed to reproduce low frequencies is commonly called a woofer. A driver optimized to reproduce high frequencies is commonly called a tweeter. A driver that reproduces frequencies in the midrange is, not surprisingly, called a midrange driver.

What is Hearing?

Hearing is the perception of sound, however that is defined. The brain interprets the information gathered by the ears, which respond to mechanical stimuli and produce electrical nerve stimulation. Humans are very good at this. We can hear a wide range of frequencies, over an enormous range of amplitude, from extremely subtle sounds to the audible results of catastrophic events.

We can hear a range of frequencies with a ratio of 1,000-to-1 or more, and perceive changes caused by frequencies beyond the (assumed) audible range of clear tones. The bones in our inner ears compress or expand the levels of sounds enabling us to hear usable information with a loudness ratio of one trillion-to-one.



We are extremely sensitive to the relationship between time and sound. We can easily detect the differences in arrival times between sounds reaching our two closely spaced ears and this is one mechanism our brains use for determining the direction of the source of sound. The time it takes for sound to decay (reverberation time) helps to define the environment we're in.

What is Sound?

Scientists would have you believe that they know the exact nature of sound and how we perceive it. If you carefully scrutinize the supposed truths they offer you'll find that many mysteries still exist.

"…many mysteries still exist."

Sound is assumed to be a series of waves composed of compressed and rarefied air (high- and low-pressure zones) but it's interesting to note that sound travels faster and farther through water, an essentially noncompressible medium. A 30Hz bass fundamental supposedly will create a sound wave 40 feet long (the distance between the pressure peaks) but we can easily hear a 30Hz signal in the cabin of a car that may be less than 6 feet in length.

One thing is fairly certain: if some force moves our eardrums inward and outward 20 or more times a second, but less than 20,000 times a second, we are likely to hear it. Loudspeaker drivers can produce that force but you'll probably need at least two—one for the lowest frequencies and one for the higher frequencies.

The signal from an audio amplifier is electrical and our ears require a mechanical force to produce eardrum movement. Loudspeakers convert electrical energy to mechanical energy through a process called transduction. This same word is used to describe the biological process of converting the mechanical motion of our eardrums into electrical impulses in our nerves.

Transduction

The process of converting an electrical signal into mechanical movement is called electromechanical conversion or transduction. The conversion device is often called a transducer.

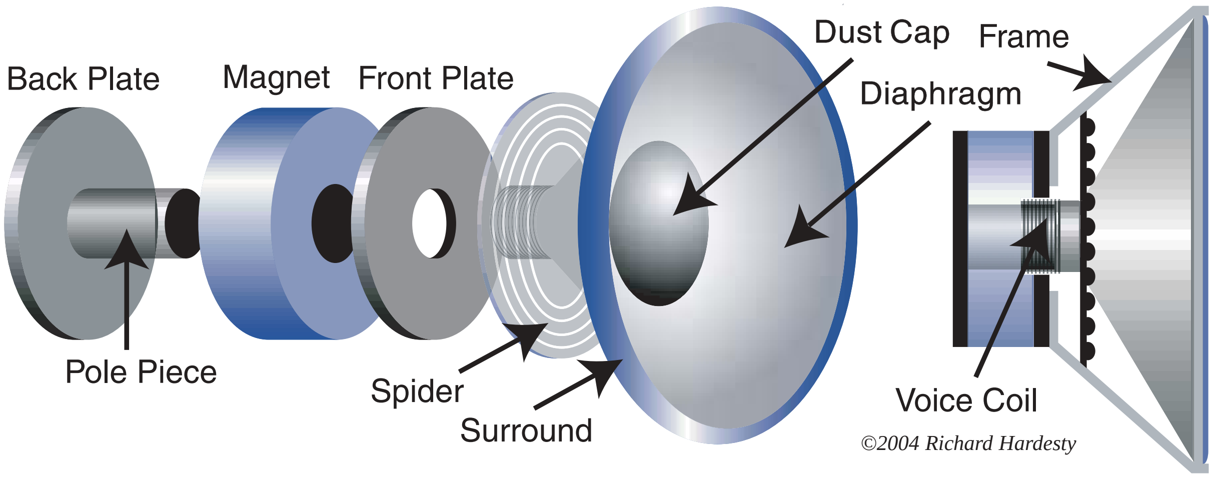
Most drive elements employ the interaction between two magnetic fields to create mechanical movement. Moving coil, planar magnetic and ribbon drivers work this way. Electrostatic drivers use electrostatic fields instead. Virtually all practical loudspeaker drivers employ either electromagnetic or electrostatic fields to move a diaphragm that stimulates the surrounding medium—which, in the case of a high fidelity audio system, is air.

Most transducers rely on the simple physical laws of attraction. Magnetic poles that are alike repel, and those which are unlike attract. Like electrostatic charges repel and unlike electrostatic charges attract. Speaker drivers rely on these principles to convert electrical energy (signal) into mechanical energy (sound).

Magnetism for Electromechanical Conversion

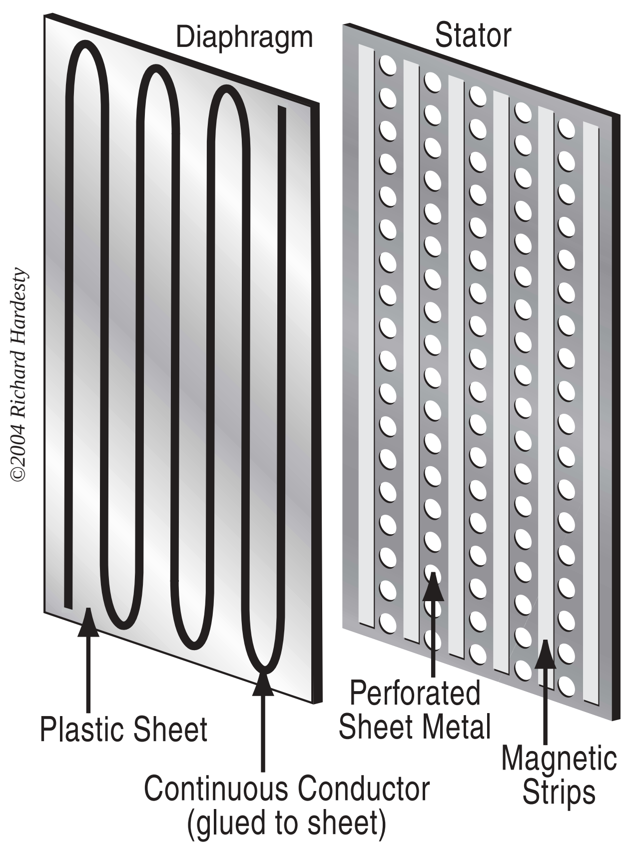
The most common means of electromechanical conversion in speakers utilizes the interaction between two magnetic fields. Moving coil (dynamic) drivers employ magnetism for energy conversion, as do planar magnetic and ribbon drivers.

A dynamic driver (see following illustration) has a permanent magnet attached to the frame with the magnetic force concentrated in a gap between the pole piece and the front plate. The voice coil is suspended in this gap and surrounded by it. The voice coil produces an electromagnetic field when fed the alternating current signal from an audio amplifier. This voice coil is attached to the diaphragm and the entire mechanism is allowed



to move back and forth within the limits of a spring suspension system, comprised of the spider and surround. (Tweeters don't usually have spiders and rely entirely on the surround for diaphragm suspension.)

A planar magnetic driver has magnets attached to a perforated sheet of metal (or other material) and a diaphragm made of a thin plastic sheet stretched across a frame. A wire or other conductor is bonded to the diaphragm and acts as a voice coil, which is energized by the signal. The magnetism induced in the



conductor (voice coil) reacts to the fields surrounding the permanent magnets attached to the chassis, and the diaphragm and voice coil (which really isn't coiled) moves back and forth within the limits of the elasticity of the diaphragm material.

In either case the signal current creates an electromagnetic field that reacts with the permanent magnetic field and moves the attached diaphragm back and forth. This mechanical movement creates sound. With a dynamic or planar magnetic driver the permanent magnet(s) is fixed and the electromagnet, energized by the audio signal, moves as the fields interact.

Dynamic Versus Planar Magnetic Drivers

Dynamic drivers have a better power-to-weight ratio because the permanent magnetic force is concentrated in a small gap and the voice coil provides an efficient electromagnet. Planar magnetic drivers rely on the permanent magnetic force radiating from the magnets themselves and the interaction of the magnetic field induced around the conductors, which are bonded to the diaphragm.

The driving force is significantly weaker in most planar magnetic designs and the highly touted reduction in moving mass is debatable. A large planar magnetic driver may have lower moving mass than a dynamic woofer but an increased moving mass when compared to a dynamic tweeter. Measurements (or listening) will show that planar magnetic drivers don't perform well at either frequency extreme. Bass performance is depreciated by partial cancellation due to dipole radiation and by excursion limitations. High frequency performance is depreciated by relatively high moving mass and relatively low magnetic forces.

Dynamic drivers can have nonlinear distortions caused by changing voice coil inductance and energy storage problems due to diaphragm resonances and chassis reflections. These faults can be largely eliminated by techniques that will be discussed in the interviews with speaker designers in this **Journal** and the one that follows.

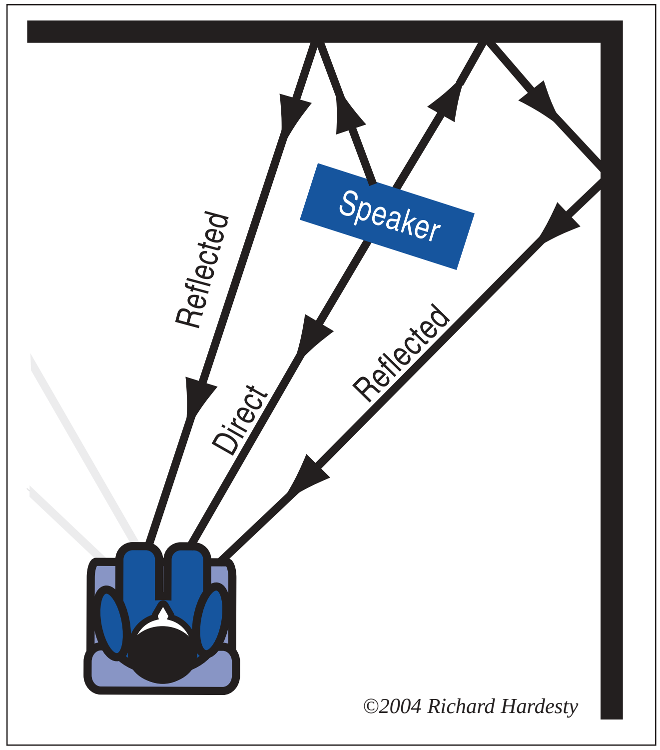
Some planar magnetic drivers are single-ended and have magnets on one side of the diaphragm only. This driver type is inherently nonlinear. As the diaphragm moves closer to the magnetic structure the magnetic force increases, and as the diaphragm moves away from the magnetic structure the force decreases. Linear travel is very limited and the large excursions required for bass are likely to cause intermodulation distortion of higher frequencies applied to the same diaphragm.

The diaphragm in any speaker that radiates from both the front and rear sides must be very large in order to produce low frequencies due to partial cancellation. Very low frequencies will be inaudible because a high-pressure zone created on one side of the diaphragm will try to rush around to fill the low-pressure zone created simultaneously on the other side.

The biggest problem with planar speakers is energy storage. The source of this time smear can be traced to the tympanic nature of an edge-clamped diaphragm and the lack of damping from an enclosure along with the rear wave radiated from an open-back source.

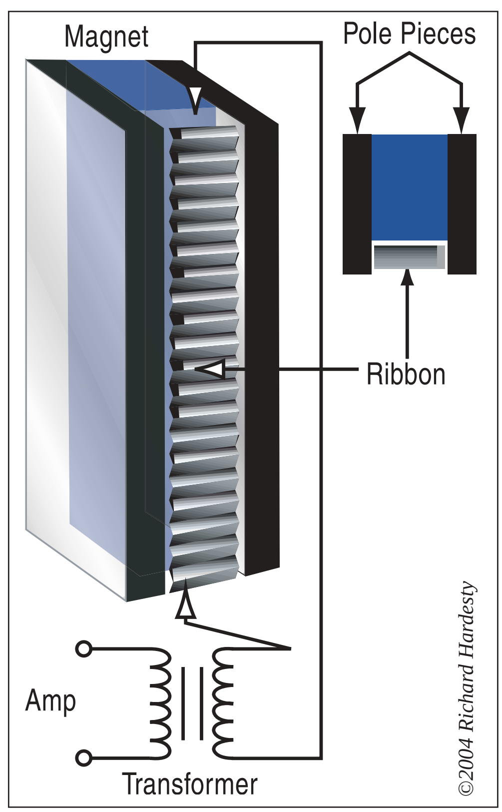
Some people are drawn to the sound of planar speakers because they are free from box resonances. Eliminating the enclosure eliminates the negative effects of panel resonances but it also eliminates the damping effect of an enclosed volume of air and allows the rear wave to escape into the room where it will reflect off the front wall and be returned to the listener delayed by an additional period of time defined by the added path length from the source to the wall and back.

This time smear may be interpreted as “openness” or “transparency,” but it’s really artificial ambience that is unrelated to the signal. Time smear actually depreciates the reproduction of real detail and some serious listeners will eventually learn to hear this artifact for what it is. Others will prefer the sound from planar speakers even though it’s not an accurate representation of the recorded signal.



Ribbons

A ribbon driver uses a conductive material, often a thin strip of aluminum, suspended within a powerful magnetic field. The conductive material is energized by the audio signal but is often coupled to this signal by a transformer to provide an acceptable impedance match. The ribbon acts as both the diaphragm and voice coil.



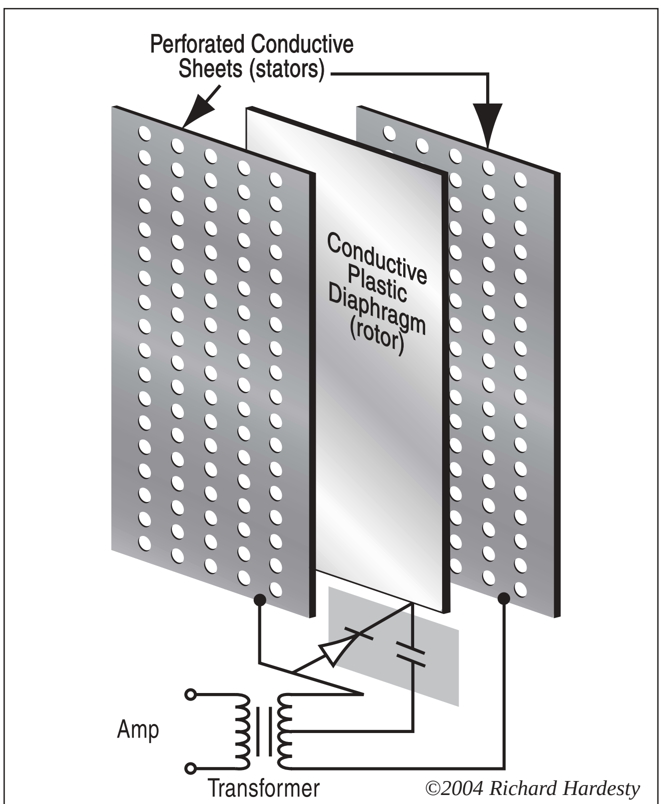
True ribbon drivers have the potential to react very quickly because of the low-mass moving element but the inherent hysteresis of the required matching transformer and the elimination of damping from behind the diaphragm negate much of this potential. It's hard to justify the claims of superiority of ribbon drivers by examining the measurements. They simply aren't demonstrably better, just different.

Many planar magnetic drivers are incorrectly called ribbon drivers. A true ribbon driver does not have an edge-clamped plastic diaphragm with a conductor bonded to it. The diaphragm is the conductor and it's usually attached only at the ends.

High Voltage for Electromechanical Conversion

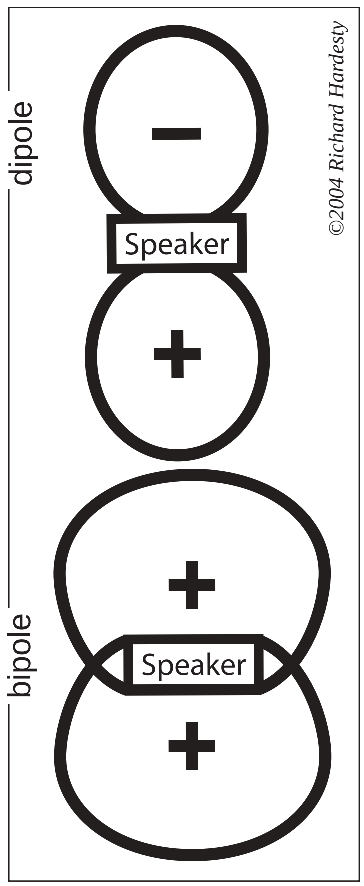
An electrostatic field surrounds a conductor that is energized with high voltage. Electrostatic fields interact much like magnetic fields and can be used to make speaker drivers that rely on relatively high levels of voltage rather than current.

An **electrostatic driver** has a conductive diaphragm suspended between two conductive, perforated plates and all three are energized with high voltage. As in an electric motor, the plates are often called stators and the diaphragm a rotor. The stators are fixed to the chassis and the diaphragm, which is typically stretched over a frame, is allowed to move within its limits of elasticity.



The signal can be used to modulate the voltage charge on the plates so that the diaphragm becomes attracted towards one plate and repulsed by the other. Much like the human dating ritual, the diaphragm will move towards the attractive plate and away from the repulsive one. When the polarity of the signal reverses, the diaphragm will move in the opposite direction.

Unlike a planar magnetic driver, the diaphragm in an electrostatic driver is driven over its entire surface rather than just at the conductors. The electrostatic diaphragm is also lighter because it doesn't have the heavier, signal-carrying conductor bonded to it. Like a planar magnetic driver, the diaphragm in an electrostatic driver is clamped at the edges causing it to perform more like a tympanic membrane than a piston.



Curved electrostatic drivers are inherently nonlinear. When the diaphragm moves forward (toward the apex of the curve) the tension increases, and when the diaphragm moves backwards the tension decreases. Curved speakers made from an array of flat segments are more linear but suffer from interference patterns sometimes derisively called the Venetian blind effect.

Electrostatic drivers are not damped by an enclosed volume of air and are usually dipoles with significant radiation towards the wall behind the speakers. Dipole radiation is an inevitable source of time smear unless there is no wall behind the speakers. Edge-clamped diaphragms are inevitable sources of time smear.

Diaphragms

The diaphragms in dynamic drivers are usually cone- or dome-shaped because these shapes provide the greatest rigidity and stiffer diaphragms are more likely to move as perfect pistons over at least some of their frequency range.

Dynamic diaphragms are commonly made from paper, treated cloth, various plastics or metal. Diaphragms made from aluminum or titanium may be anodized, creating what is advertised as "ceramic coated" material, or hard anodized, creating what is advertised as "ceramic." Carbon fibers are often added to damp resonances and various materials may be sandwiched together for the same purpose.

Planar magnetic and electrostatic drivers use thin plastic film with conductive materials bonded to the film. A diaphragm in an electrostatic driver may simply be painted with a conductive material allowing it to be charged up with the high voltage required to create an electrostatic field. The diaphragm in a planar magnetic speaker will usually have a wire or flat aluminum conductor bonded to the film to act as a voice coil. This conductor must be sufficiently robust to carry the audio signal but light enough to allow some high frequency response. A planar magnetic diaphragm of the same size will be significantly heavier than an electrostatic diaphragm. Because the diaphragm in a planar magnetic speaker is usually significantly larger it may be significantly heavier than the moving mass found in a dynamic dome tweeter.

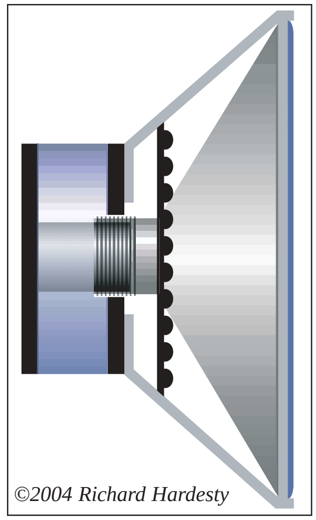
The diaphragm in a planar magnetic or electrostatic driver is usually stretched across a frame preventing it from acting in the desirable piston-like manner. A diaphragm that is clamped around the periphery cannot move back and forth as a unified entity. It acts instead like a tympanic membrane storing and releasing energy over an extended period of time.

Visualize this effect as ripples on the surface of water in a large tub. If a rock is dropped into this tub ripples will emanate outward from the point of entry. When the ripples reach the edge they will be reflected back and will form interference patterns with the originals.

Like a tympanic membrane, a stretched diaphragm will usually exhibit a prominent resonance in the lower midrange, or at a higher frequency if the diaphragm is smaller.

Ribbon drivers use a thin strip of conductive material, which acts as both the diaphragm and voice coil. This material is often accordion shaped and attached to the driver frame only at the ends. This is a fragile arrangement but it allows the diaphragm to behave more like a piston than a tympanic membrane. Unfortunately it also presents very low impedance to current flow, usually requiring a matching transformer to present an acceptable load to the amplifier.

Woofers



A woofer is a driver engineered to reproduce low frequencies. Its diaphragm must be either very large or allowed to travel back and forth over a substantial range in order to deliver deep bass frequencies at audible levels. Two sources of distortion tend to keep dynamic bass drivers within reasonable size limits.

A very large diaphragm will "break up" and fail to perform well at higher frequencies, where it will cease to act as a piston and start to behave like many smaller drivers moving at random rather than in unison with the voice coil. Distortion will also rise with an increase in diaphragm travel, due to various nonlinearities, preventing small diaphragms from delivering ideal performance at low bass frequencies.

Drivers with dipole radiation patterns (open back) will suffer from cancellation at bass frequencies. Planar woofers will need very large diaphragms in order to produce a semblance of bass and simply can't deliver very low frequencies with impact.

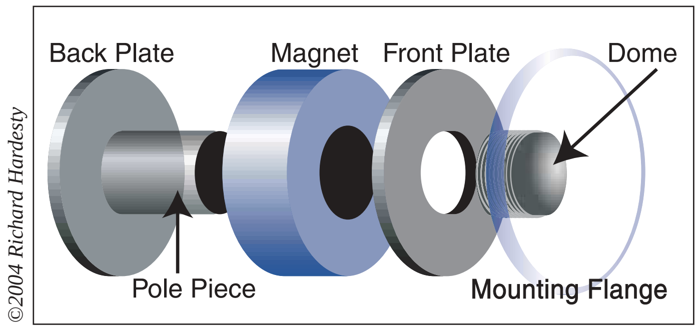
Tweeters

A tweeter is a driver engineered to reproduce high frequencies. Its diaphragm must be quite small so that it can accelerate and stop rapidly in order to accurately replicate higher frequencies, and so that it will disperse high frequencies over a wide area.

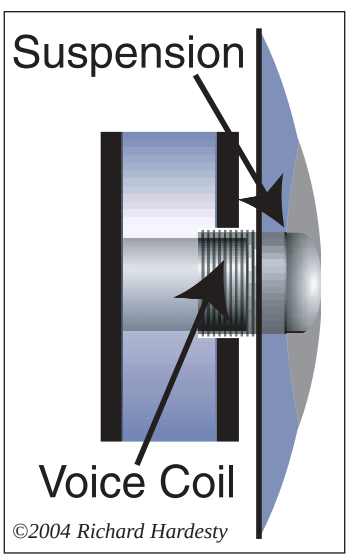
A tweeter diaphragm will generally be one inch in diameter or less. A larger diaphragm will extend the lower end of the driver's response at the expense of the higher end, and will result in a speaker system with less uniform polar response (dispersion pattern).

"Tweeters with diaphragms made from various metal alloys can offer response that extends to 30kHz and beyond."

Tweeters with soft diaphragms will start to break up (perform in a nonpiston-like manner) at about 10kHz. Virtually all manufacturers with access to laser-interferometry equipment realized



this fact decades ago and converted to diaphragms made from stiffer materials. Some think soft dome tweeters sound more musical but I believe that they are being seduced by the time smear that results from a diaphragm in completely incongruous motion (unrelated to voice coil movement).



Tweeters with diaphragms made from various metal alloys can offer response that extends to 30kHz and beyond. I think this extended response is necessary for accurate musical reproduction.

There are valid arguments supporting ribbon or electrostatic tweeter drivers but I've never heard an effective blend with planar tweeters and dynamic low frequency drivers.

Midrange Drivers

A midrange driver is used to reproduce the critical frequencies in the midrange and to balance the radiation pattern of the loudspeaker. The driver needs to be small enough to produce midrange detail with wide dispersion up to its higher frequency limits. It needs to be robust enough to provide adequate output levels.

"A 6- to 7-inch driver is a woofer, not a midrange..."

The best midrange drivers are 3 to 4 inches in diameter in my opinion. A 6- to 7-inch driver is a woofer, not a midrange, and when a driver this big is used for midrange frequencies, detail suffers and the polar pattern (dispersion) gets ragged. A speaker system with an uneven pattern of dispersion will be very sensitive to placement because the energy reflected from the room boundaries, particularly the side walls, will have a different tonal balance than the direct energy radiated toward the listener.

Horns

Conventional dynamic drivers may be horn-loaded or a compression driver may be used with a horn. Compression drivers and horns are commonly found in sound reinforcement systems but have no place in high fidelity audio in my opinion.

Compression drivers are inherently nonlinear and their only valuable attribute is high sensitivity. Horns have been studied for decades and their distortion characteristics are well known. In my opinion, their only attribute is sensitivity. Directivity is a desirable characteristic for sound reinforcement where both compression drivers and horns are used effectively but has little value in the living room.

A speaker system using horn-loaded drivers can be driven by a 3-watt, single-ended, triode amplifier. You may find this combination desirable but I don't. I'd rather use a speaker system based on more accurate drivers with less coloration, and drive it with a readily available amplifier that produces more power. The results will be every bit as dynamic but more reliable and, more importantly, will sound better! APJ

How Crossovers Work

by Richard Hardesty

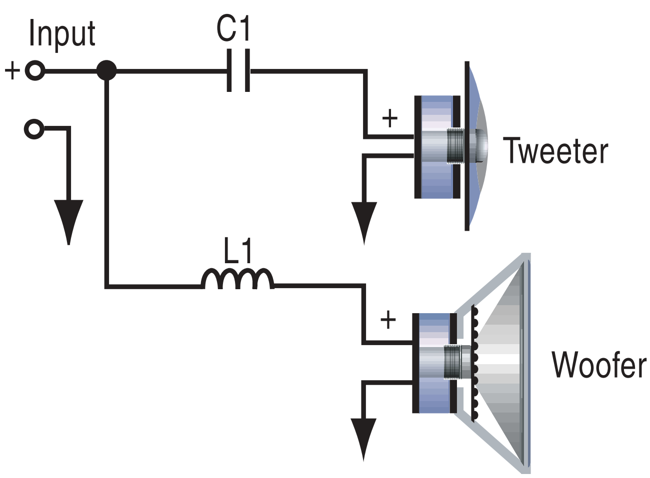
A crossover or dividing network is used to separate the audio signal into various bands of frequencies and direct each band to the appropriate drive element within a loudspeaker. A crossover network is comprised of various filter networks, which may be described as high-pass, low-pass or band-pass (a combination of high- and low-pass filters).

Bands

The pass band is a range of frequencies that is allowed to pass to a specific driver relatively unimpeded. Frequencies that fall outside this range are presented with increasingly rising impedance, which attenuates rather than blocks the frequencies that fall in the stop band(s)—the range of frequencies outside the pass band.

A filter system handling the midrange might provide a pass band of 500Hz to 4,500Hz, which would be an acceptable range of frequencies for a midrange driver. Frequencies below 500Hz and above 4,500Hz would fall into the stop bands of this network, which would consist of a 500Hz high-pass filter combined with a 4,500Hz low-pass filter, and would be presented with rising impedance. The impedance would rise at a rate commensurate with the slope of the filters.

First Order Crossover

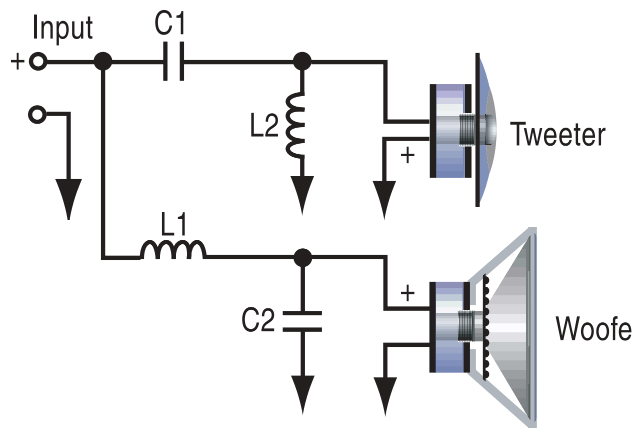


Common crossovers contain filters, which are designated as first- to forth-order, specifying the slope of attenuation. A first-order slope is 6dB/octave and they increase in 6dB increments from there. A second-order slope is 12dB/octave and so forth. A third-order slope is 18dB/octave and a forth-order filter provides 24dB/octave slopes in the stop band.

Slope

Slope refers to the rising impedance to current flow and the commensurate reduction in driver output. Frequencies in the stop band encounter rising impedance, not a wall. These frequencies are attenuated, not actually stopped. As the impedance rises, current flow is reduced and less current flows to the drive element that is fed by the network. Reduced current results in reduced output, which means less sound at the frequencies that are attenuated. The quicker the impedance rises the steeper the slope.

Second Order Crossover



Phase Shift

A filter network will induce a phase shift of about 90 degrees in the stop bands for each order of magnitude. This is a simplistic statement meant to illustrate the problem, not provide engineering instruction. Voltage will lead current through a primarily inductive filter and current will lead voltage through a filter that is primarily capacitive.

Active Networks

Active crossover networks utilize active components like transistors, which require AC power. Active networks work on the line level signal that precedes the amplifiers. The line level signal following an active crossover network, which has been divided into frequency bands, must be amplified. The separate bands are amplified independently creating a biamped or tri-amped (etc.) speaker system.

Active networks have less insertion loss and are often used in sound reinforcement speakers but are seldom found in high fidelity speakers. High-end audio speakers generally employ passive crossover networks, which allow impedance compensation and frequency response correction tailored to the specific drive elements used in that speaker.

"…a crossover is an important part of a loudspeaker…"

Passive Networks

Passive crossover networks utilize passive components like resistors, and reactive components like capacitors and inductors. These components work on the speaker level audio signal and don't require external power.

A passive crossover network can tailor the response of each driver to eliminate frequency peaks or dips and impedance fluctuations allowing flat response with a wide variety of amplifiers.

Capacitors

A capacitor blocks DC and provides impedance to current flow that rises in inverse proportion to frequency as the component becomes reactive. The frequency where the capacitor becomes reactive is determined by its time constant and defined by its value.

Capacitors are used as high-pass filters because they impede the flow of lower frequencies starting at a frequency determined by their value. Capacitors are represented by a symbol that shows their construction: two plates divided by a dielectric material with a wire attached to each plate.

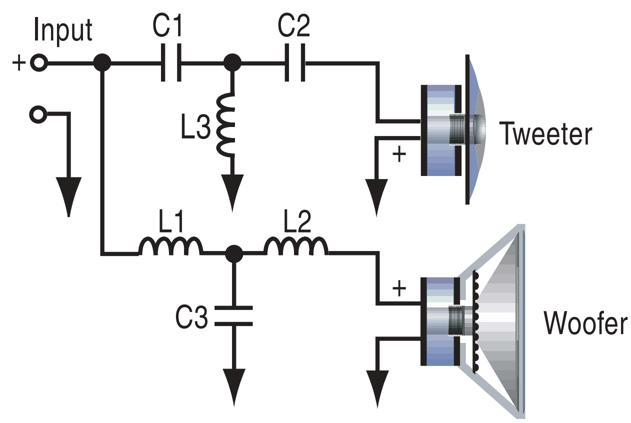
A single capacitor in series with the tweeter will act as a first-order high-pass filter and attenuate frequencies below the crossover point at a rate of 6dB/octave. The crossover point will be established by the impedance of the tweeter and the value of the capacitor.

Steeper slopes are attained by using multiple reactive components. A capacitor in series with a tweeter followed by an inductor in parallel (shunt) will provide a 12dB/octave (second-order) high-pass filter.

Inductors

An inductor is a coil of wire with or without an iron core. Inductors with air cores are represented by a continuous line with humps (rounded wiggles) to indicate varying impedance to

Third Order Crossover



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current flow. An iron core inductor will have one or two straight lines below the wiggly line to indicate the presence of the core.

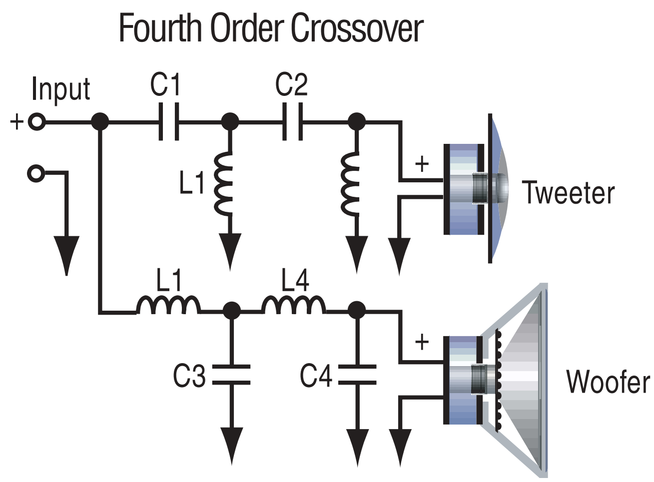
Inductors store and release energy through the creation and collapse of a magnetic field. Inductors provide impedance that rises with frequency above the frequency where the inductor becomes reactive. They are used as low-pass filters and placed in series with drivers to attenuate high frequency information.

A single inductor in series with a woofer will act as a 6dB/octave low-pass filter. Steeper slopes can be achieved by using additional components. An inductor in series with a woofer followed by a capacitor in parallel (shunt) will provide a 12dB/octave (second-order) low-pass filter.

Iron core inductors can provide more inductive reactance with less wire and therefore have lower DC resistance. Reducing DC resistance can increase speaker efficiency but iron cores can saturate and create distortion. Iron core inductors are seldom found in high quality speakers except when used only for very low frequencies.

Resistors

A resistor is represented by a line which zigzags to suggest a component that resists current flow. A resistor impedes the flow of current at a constant rate regardless of frequency. Resistors are used to reduce driver output in order to balance a speaker that utilizes multiple drive elements. Resistors will also be found in shunt elements that adjust impedance curves and driver response irregularities.



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Assembled Networks

A crossover network is an assembly of high- and low-pass filters that tailor the response of the loudspeaker. A 3-way speaker, for instance, might have a low-pass filter preceding the woofer so that it will only reproduce bass frequencies; a high-pass and a low-pass filter preceding the midrange to limit its response to midrange frequencies; and a high-pass filter preceding the tweeter to limit its response to high frequencies exclusively.

Other elements in the complete crossover network will adjust for impedance variations and driver response irregularities as well as level differences between the various drive elements. To say that a crossover is an important part of a loudspeaker is an understatement. APJ

How Speakers Work

by Richard Hardesty

A loudspeaker is assembled from drivers, a crossover (dividing) network, and an enclosure that provides the mechanical support and acoustic loading for the drivers. A complete speaker system provides electromechanical conversion so you can hear the electrical signal that an audio system creates, and offers an acceptable esthetic design for use in the home.

Each driver makes sound over a limited range of frequencies. This range is determined by the crossover network, which directs the proper band of frequencies to the appropriate drive element, and the acoustic loading provided by the enclosure which tailors the response of individual drivers.

The complete loudspeaker unit should provide an accurate acoustic replica of the electrical input signal, and be small enough to fit comfortably into the domestic environment. It should be sensitive enough to allow operation with readily available amplifiers and should play loudly enough to provide acceptable dynamic range.

A stereo speaker system will include at least two loudspeakers. A surround sound speaker system will include additional units, often totaling six or more. There is no reason that a surround sound speaker system shouldn't be held to the same performance requirements. An individual speaker is either a high fidelity transducer or it isn't.

Bandwidth and Frequency Response

A speaker system that qualifies as an acceptable entry-level high fidelity transducer should have frequency response of at least 40-15,000Hz and this response should be accurate to within ±3dB from about 50Hz upward. A speaker system that doesn't meet these requirements can't possibly reproduce music recordings accurately. But a speaker system that does meet these requirements won't necessarily sound good because there are additional requisites for good sound.
The following statement may be the most important in this

Journal: Measured accuracy is required for good sound but measured accuracy alone doesn't guarantee good sound. The opposite is true: Measured inaccuracy assures that accurate performance is impossible.

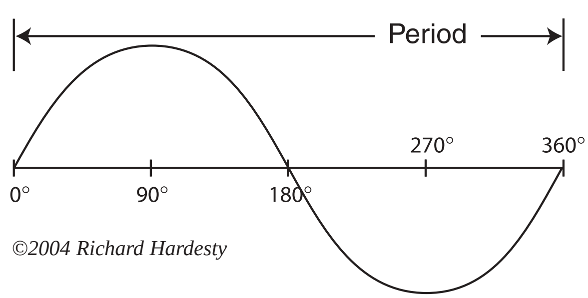
Flat frequency response is the cornerstone of accurate reproduction. If some frequencies play louder than others, some notes (or parts of notes called harmonics) will be louder than others. Accurate reproduction of musical timbre requires accurate frequency response (within close limits) over a wide range. Accurate reproduction of musical rhythm and pace (called tempo in music) requires accurate frequency response (within reasonable limits). Period.

Personally I require a speaker system with useful response from about 10Hz to about 30kHz. I demand a speaker system that provides time- and phase-accuracy so that the output waveform is a respectable facsimile of the input signal. These requirements are easy to talk about but very hard to achieve.

Phase

Phase is a way of expressing the time relationships between frequencies, which are usually depicted as waves, which can be divided into sections. A complete wave starts at a reference

SINE WAVE ILLUSTRATION



line (0), goes to a positive peak and back to the reference line, then goes to a negative peak and back to the reference line. This complete waveform, called a period or cycle, can be divided into 360 degrees. The instant when a second wave starts or stops can be expressed in terms of phase lead or phase lag relative to the first wave, describing their relationship in time.

There are two primary factors that affect the time domain accuracy of a loudspeaker: phase shift caused by crossover components, and the physical position of the drive elements, which is dictated primarily by the rise-time of individual drivers. A time- and phase-accurate speaker requires careful attention to both factors.

> ***"Measured accuracy is required for good sound but measured accuracy alone doesn't guarantee good sound."***

The natural sounds of voices and instruments are made up of fundamental tones and harmonic overtones. The combination of several sound waves results in the unique timbre of the voice or instrument. We perceive the unique timbre of the resultant combination of sound waves, which allows us to identify the voice or instrument that produced the fundamental tone.

A piano and a violin can play the same note at exactly the same fundamental frequency. They sound different because each instrument produces a combination of frequencies made up of the fundamental (same frequency) and a unique set of harmonics (different combinations of overtones). This combination and balance of frequencies create the character or color (timbre) of the note. The same is true of the human voice.

You can easily identify your sister's voice—no one else sounds exactly the same. She may be able to sing the same note that the piano and violin play. If the voice and each instrument produce the correct note, the fundamental frequency will be identical. Each of the three entities will sound different because of a unique set of harmonics at various frequencies that provide timbre.

The time relationships between the various frequencies that are perceived as a musical note with distinct timbre are critical in real life and are considered important in virtually all audio components except speakers. Why? Because it's hard to make time- and phase-accurate speakers and many manufacturers will simply tell you that it doesn't matter. They're wrong.

A speaker with a third- or fourth-order transition between drive elements cannot produce an output waveform that resembles the input signal, no matter how the drivers are positioned. Phase shift is caused by the crossover components and varies with frequency. You can't simply realign the drivers to correct for this problem—it's a moving target.

A speaker with third-order filters will typically have every other driver wired out-of-phase to enable flat frequency response. The phase shift produced by the crossover components will bring the drivers back into phase in the transition (overlap) range preventing amplitude cancellation. The drivers will be out-of-phase in their pass bands.

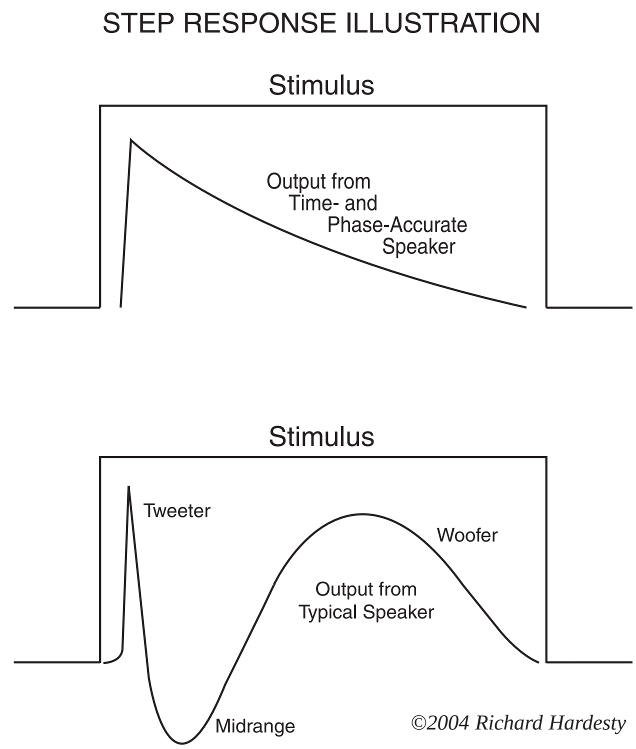
"Flat frequency response is the cornerstone of accurate reproduction."

A 3-way speaker with third-order filters will have a midrange driver that is out-of-phase with the woofer and the tweeter. The woofer will push while the midrange pulls. A note with a fundamental that occurs at the upper end of the woofer pass band may have the first few harmonic overtones reproduced by the midrange driver in reverse phase. The upper harmonics may be reproduced back in phase with the woofer.

This is clearly audible as an alteration of timbre and image and I have demonstrated this fact to thousands of people. It may also alter the perception of musical rhythm and pace and dynamic range. Phase really gets to be a nightmare when combining 3-way and 2-way speakers with different crossover points in a surround sound system.

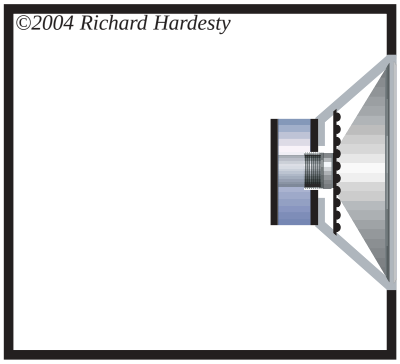
A 3-way speaker with a fourth-order network has a crossover with about 360 degrees of phase shift. A network can be designed that allows all drivers to be wired in phase but the midrange driver in such a speaker will be out-of-time by an entire cycle due to a complete rotation of phase (360 degrees).

How do you tell if a speaker is time- and phase-accurate? By observing the step response graph. The step response stimulus resembles half of a square wave. A time- and phase-accurate speaker will produce a triangular output in a positive direction (above the reference line). A speaker can't reproduce the flat line at the top because a speaker can't reproduce the DC portion of the step. (A speaker only makes sound when it's moving.) Most loudspeakers, which are not time- and phase-accurate, will produce an output that is scrambled in time and phase. See the illustrations below.

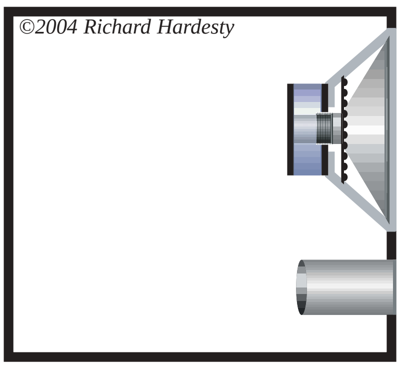


Bass Loading

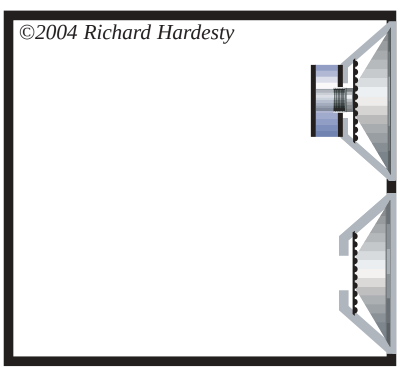
Dynamic bass drivers may utilize different loading techniques. Loading is required because the driver diaphragm creates two sound waves in opposite directions. These waves are out-of-phase and may cancel if they are not isolated from each other. The wave that comes from the front of the diaphragm is called the front wave and the wave that emanates from the rear of the diaphragm is called the rear wave.



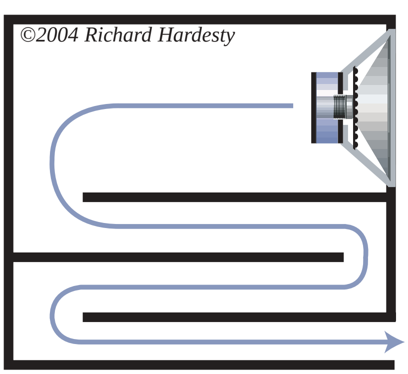
SEALED ENCLOSURE



VENTED ENCLOSURE



PASSIVE RADIATOR



TRANSMISSION LINE

The front wave may radiate directly towards the listener or that radiation may reach the listener indirectly. The rear wave may be completely isolated from the front wave by a sealed enclosure or the rear wave may be used to augment the front wave by a vented (ported) enclosure tuned to resonate at a certain frequency.

A vented enclosure uses the resonating volume of air in a port (vent) to augment bass output and minimize driver excursion at the resonant frequency. This technique can be effectively employed to extend the bass response of a full-range speaker that utilizes the bass driver to also reproduce part of the midrange. A vented speaker will ring twice as long as a sealed design (after the signal stops) and will tend to emphasize the vent resonant frequency.

A true transmission line may be used to delay the rear wave until it can be added to the front wave in phase. Even in theory this can only work over a narrow range of frequencies and the line needs to be about one-quarter of the target wavelength. A speaker with a true transmission line needs to be quite large if low bass is to be reproduced.

I have heard every engineering compromise possible and built many for my personal edification. My subjective impressions are pretty simple: low bass sounds best from a sealed enclosure. Vented enclosures can be used with a reasonable compromise in performance in full-range speakers but have no place in subwoofers designed for music. Transmission lines work well to manipulate impedance but smear bass frequencies over time and deliver bass with a character that I find less than satisfactory.

You may come to different conclusions and your ears should be your guide. Read the literature carefully as you listen to make sure you know what you are listening to. A passive radiator is a vent substitute and a system that utilizes a passive radiator can be considered along with conventional vented enclosures.

Impedance Effects of Bass Loading

A sealed enclosure will have a single resonance and a single impedance rise that coincides with this resonant point. A vented enclosure will have two impedance peaks that straddle the vent resonance. A well-designed transition line will have a well-damped impedance rise or none at all.

A self-amplified and equalized woofer system can have no resonance in the pass band and no impedance rise. Examples include the Bag End and Vandersteen subwoofers, and the Vandersteen 5A speakers reviewed in this **Journal**.

Dispersion Pattern

The dispersion pattern, or polar response, of a loudspeaker will have a substantial effect on how that speaker sounds in a room. A speaker with a broad and uniform dispersion pattern will produce reflected waves from the room boundaries that have a similar tonal balance to the direct radiation from the speaker. This requires drivers that have a small diameter relative to the wavelengths produced and usually means that a 3-way or 4-way speaker system will be necessary.

There are many speakers in today's market that are actually 2-way designs with augmented bass. These have "midrange" drivers, which are actually woofers and are too large to reproduce fine details or deliver wide dispersion in the midrange. These speakers will be very critical to positioning because the reflections from the side walls will have a completely different tonal balance than the direct radiation from the speaker. Some will attribute this problem to incredible accuracy but it's actually a design fault.

Planar speakers are seductive to many listeners but they simply can't be called accurate by even the least strict definition. Planars can't produce any real bass due to partial cancellation, so many are hybrid designs. Some try to combine dynamic woofers with electrostatic or ribbon tweeters. Some try to position woofers and tweeters side by side. Most allow the rear wave to radiate into the room and smear the signal over time.

Sound pressure from a line-source, which many planar speakers try to emulate, falls linearly with distance. Sound pressure from a point-source, which all woofers become at low frequencies, falls with the square of distance. A hybrid speaker with a point-source woofer and a line-source tweeter will be in balance at only one listening distance. Reflected waves will have a different tonal balance.

A speaker with the tweeter beside the woofer will deliver a different tonal balance to every listener in the room because a small lateral movement will change the relative distance between the listener and each of the drive elements. The reflected waves, like the splash off the wall behind the speakers that occurs with all dipolar radiators, will have an entirely different tonal balance.

Baffles

Most dynamic speakers are mounted on a baffle. A baffle changes the frequency response of the drivers and the sound of the speaker. Baffle reflections are a major source of time smear and these reflections are delayed and distorted signals that interfere with the signal that emanates directly from the drive element.

A speaker with a baffle the size of a refrigerator will produce a sound that is closed down and confined even if the enclosure is completely dead. You can simulate the effect by holding up two album covers, one on either side of your mouth while speaking. Have someone listen to your voice with and without the "baffles" and observe the difference.

Even if the enclosure is made from fairy dust by skilled craftsmen from Mars, a speaker with a large baffle will sound like a big box because that's what it is.

Nonresonant enclosures are admirable but can't make up for average-quality drivers or large baffle surfaces. APJ

Segue

You can learn a lot about how a complete loudspeaker functions by examining an outstanding product and evaluating its component parts. The following review of the Vandersteen Model 5A explains the speaker system that I chose to purchase for my own use and makes the reasons for my choice obvious.

This review shows how the Vandersteens differ from alternative products and mentions some of the speakers that I used for comparison.

Vandersteen 5A Review

by Richard Hardesty

The Model 5A speaker system was developed over a period of nearly thirty years and evolved from previous Vandersteen systems that had been refined to their performance limits. This is not simply advertising hype—I sold Vandersteen speakers from the beginning and I was there to watch the innovations occur and the products progress.

This review is a first-person report with observations from someone (me) who is intimately familiar with Vandersteen *speakers and virtually all the other top brands developed over the last several decades. I sold, installed, repaired and compared all the major competitors. Many speakers were used in my home system over extended periods so that I could become thoroughly familiar with them.*

My retail store was an authorized dealer (at one time or another) for Magneplanar, Audire and Eminent Technology planar magnetic speakers—Acoustat, Quad, Sound Lab, Dayton-Wright and Martin-Logan electrostatic speakers—Decca and Sequerra ribbon speakers—Thiel, KEF, B&W, Dahlquist, Gale, Braun, Mirage, Snell, Ohm, Spica, and many other dynamic speakers. Within this mix, Vandersteens were always leading contenders for top performance and were unrivaled for value.

Evolution not Revolution


3A Signature

They didn't have to reinvent the wheel to produce the Model 5As. Since the first commercial products were delivered, all Vandersteen *Aligned Dynamic Design* speakers have embodied certain principles. Good sound was important, of course, and Vandersteen speakers were required to accurately replicate the signal, just like other high fidelity audio components. This may seem like a joke but it's not.

Few people would accept an amplifier that delivered midrange frequencies out-of-phase with the bass and treble frequencies, but that's exactly what most speakers do. While some speakers are capable of providing fairly accurate amplitude response, phase relationships are often ignored.

All Vandersteen models have flat frequency response within narrow limits of error and exhibit excellent performance in the time domain. Flat frequency response assures that the speakers are free from euphonic colorations, and accurate time-domain performance allows the speakers to produce an acoustic replica of the electrical input signal—a task that remains unachievable by most speaker systems sold today.

The original goals of Vandersteen Audio haven't changed. The products have been refined and improved again and again as better parts have become available and new facts have been observed.

Preserving Phase Integrity

The correct reproduction of musical timbre requires accurate frequency response and precise timing and that requires accurately replicating both the amplitude and phase of the input signal. Delivering a properly focused, three-dimensional image is also dependent on correct time-domain performance, a fact that can be easily demonstrated.

"All drivers are wired in phase and the crossover networks are designed to keep them in phase."

Producing an acoustic output that correctly follows the electrical input signal requires time- and phase-accurate speakers with a minimum of energy storage (time smear). Three major sources of energy storage which smear transients over time are resonances, ringing and reflections. Vandersteen speakers are engineered to produce fewer of these time smear aberrations while maintaining the phase integrity of the original signal. All drivers are wired in phase and the crossover networks are designed to keep them in phase.

In addition to first-order acoustic transitions between drive elements and temporal alignment of the drivers, Vandersteen speakers incorporate many innovative methods to eliminate resonances, reflections and stored energy, as well as other causes of time smear. Minimum-baffle enclosures and patented reflection-free midrange drivers, along with resistive acoustic transmission lines that absorb rather than reflect rear waves are among the unique attributes of Vandersteen speakers. Damped high-loss materials are used in the construction of driver diaphragms and enclosure panels to further reduce resonances and mechanical energy storage.

Many design breakthroughs originated at Vandersteen Audio. These innovations were quietly incorporated into a relatively inexpensive line of products, which became known primarily for value.

First Designs Based on FFT Analysis

Vandersteen pioneered the use of FFT (fast Fourier transform) computer analysis years before other speaker manufacturers started to use this technology. Before MLSSA and CLIO there was General Radio.

The General Radio #2512 instrument was among the first of this product type to reach the market. Vandersteen started examining speaker performance with this exceptional device in the 1970s and the GenRad's 100kHz bandwidth keeps it in the forefront of technology even today. Vandersteen also uses the industry standard MLSSA measuring system for both quality control and hand adjustment in the anechoic chamber.

> ***"Vandersteen pioneered the use of FFT (fast Fourier transform) computer analysis…"***

Richard Vandersteen trained with the late Richard Heyser at Cal Tech on the use of the (then) newly introduced Crown (now Gold Line) TEF (time, energy, frequency) analyzer.

Time- and phase-accurate speakers were developed as a direct result of this research and the time smearing effects of energy storage and reflection became clearly evident for the first time. The use of computers and the Fourier transform changed loudspeakers forever by providing designers with insights that couldn't be obtained any other way.

Today, new Vandersteen designs are evaluated utilizing these techniques to assure accuracy, and newly manufactured products are measured to assure consistent performance.

Eliminating Reflections

Vandersteen's time- and phase-accurate designs have always provided a more detailed and transparent sound than planar speakers. The "boxless" minimum-baffle configuration produced a more open and spacious sound with better transient response than conventional speakers, which were hampered by baffle distortions. Control of diffraction and reflected energy was of paramount concern from the beginning.

> ***"The 'boxless' minimum-baffle configuration produced a more open and spacious sound"***

The introduction of the patented reflection-free midrange driver in the 1980s erased the last vestige of planar superiority by eliminating the early reflections common to dynamic midrange drivers.

Versions of this driver first appeared in the Vandersteen Model 4 and later in the Model 3. The standard version is available today in the 3A Signature speakers and the Model 5. The latest version, with an entirely new woven diaphragm, comes only in the Model 5A.

Resonance-Free Subwoofers

In the early eighties, when Vandersteen developed the first aperiodic subwoofer that accomplished a first-order transition to the main speakers, many dealers (including me) discovered the benefits of active subwoofers and passive high-pass filters.

The 2W subwoofer system utilized a first-order, passive high-pass filter that was inserted in front of the main amplifier to roll-off the bass response from the main amplifier and speakers. Low frequencies were handled by specialized amplifiers in the subwoofer, which were optimized for the job and the load. Bass amplifiers sampled the output from the main amplifier to provide a perfect blend between the main speakers and the subwoofer(s). This resulted in amazing performance gains and subwoofer integration previously unknown.

"First aperiodic, first-order subwoofers..."

The 2W was aperiodic (had no response resonance) within its pass band and used slot-loading to equalize the pressures on each side of the diaphragms and to provide a predictable interface with the room. Slot-loading also made a first-order transition possible without coloring the midrange.

Feed-forward error correction and a unique "preview circuit," which analyzed the incoming signal and modified the output preventing electrical or mechanical misbehavior, were introduced in the 2W subwoofer. Vandersteen subwoofers simply can't be overdriven and won't make unseemly noises regardless of the signal they are fed.

The 2Wq subwoofer introduced a control to vary the "Q" of the device in order to adjust the contour of response. The introduction of the 2Wq gave users greater latitude in subwoofer placement allowing the main speakers to be positioned for best image and the subwoofers to be positioned for best bass response.

Better Bass, Better Integration

Vandersteen proved that using a specialized amplifier for bass frequencies, thus relieving the main amplifier and speakers of this difficult chore, could greatly reduce intermodulation distortion and increase dynamic range providing vastly improved bass and midrange performance. These subwoofers were designed for music not sound effects.

Years before home theater was conceived, thousands of music listeners enjoyed enhanced satisfaction by combining Vandersteen 2 speakers with a 2W subwoofer or Vandersteen 3A speakers with one or two 2Wq subwoofers. These wise buyers got vastly improved performance and spent a lot less money than less thoughtful consumers. These high-value, ultra-high-performance speaker systems were hard acts to follow but improvement was still possible for those with bigger budgets.

Battery-Biased Crossovers

Vandersteen was the first to offer crossover networks in commercial speaker systems with battery-biased film capacitors. A constant charge on the capacitor dielectrics allows the speakers to sound as good when first turned on as they do after several hours (or days) of playing.

This technology has since been applied to cable dielectrics and it works very well. There are batteries all over my audio system and this advance stems from the first Vandersteen Model 5 speakers.

Adding it all Up

The Vandersteen Model 5 speakers were introduced in 1997 and provided an evolutionary improvement in performance and a new benchmark in value. They combined all the features of their high-value predecessors and added a fine-furniture appearance. Continuous refinement in virtually every area has resulted in the Model 5As.

The Model 5A system embodies every performance breakthrough that Vandersteen has produced over nearly three decades of research and development and represents the culmination of everything the company has learned about making truly accurate speakers. The Model 5A is a Model 5 with a new power supply for the bass amplifier, improved midrange and tweeter drivers and even more refined sound. The Model 5A speakers advance the performance standards set by the systems that preceded them and set new standards for cosmetic elegance.

Product Description

The Model 5A is a 5-driver, 4-way, time- and phase-accurate speaker system that incorporates a unique aperiodic, push-pull, powered sub-bass system using the most sophisticated driver made, a battery-biased passive high-pass filter and an 11-band equalizer—in a Vandersteen "boxless" minimum-baffle, nonresonant enclosure system.

The Model 5A utilizes a transmission-line loaded mid-bass driver with a proprietary Kevlar®/poly laminated cone—the patented Vandersteen "reflection-free" midrange driver with a linear surround and a tri-material, woven diaphragm—and a unique Vandersteen multichambered, ceramic coated alloy dome tweeter with an investment-cast chassis that is hand-adjusted to eliminate virtually all resonances to 30kHz and beyond. The midrange and tweeter drivers are also loaded with terminated, resistive transmission lines and feature ferrofluid-cooled voice coils.

All drivers operate in phase and are temporally aligned. The system is integrated with a completely balanced, battery-biased crossover network that provides first-order acoustic slopes to assure accurate waveform reproduction. There is a rearward-facing auxiliary tweeter that is only used in very large or very dead rooms to compensate for excessive absorbent material or the absence of reflective surfaces.

There is far more technology incorporated within this speaker than is available anywhere else at any price. I'll try to describe it piece by piece as space and knowledge permit.

Sub-Bass System

The Vandersteen subwoofer system takes advantage of the following facts. A low bass driver in a sealed enclosure has very predictable response characteristics. An internally amplified and equalized bass system can correct for any deviations (roll-off) or aberrations (response errors) and provide flat frequency response and impedance to well below the range of human hearing. The additional roll-off created by a passive high-pass filter in front of the main amplifier and speaker can be easily compensated for as well.

> ***"There is far more technology incorporated within this speaker than is available anywhere else at any price."***

Slot-loading the front wave from the bass driver equalizes the pressures that the diaphragm will encounter as it moves back and forth and allows predictable coupling to the room. (The slot is placed at floor level facing to the rear.) An indirect radiation path from the front of the driver also helps to absorb any out-of-band frequencies that might sneak past a first-order low-pass filter, preventing midrange coloration.

Some nonlinearities may occur in dynamic drive elements due to changes in inductance as the voice coil moves inward (encountering more iron) and outward (encountering less iron). Copper rings and shaped pole pieces can help but creating a complete push-pull bass system can eliminate virtually all sources of distortion.

Amplifiers that operate only below 100Hz can be optimized for performance in the low bass region. Specialized bass amplification can be tailored to the exact characteristics of the drive elements for exceptionally refined yet powerful bass that extends to infrasonic frequencies. Feed-forward error correction can assure that this bass is extraordinarily accurate to well below the range of human hearing.

So how is all this accomplished? By skillful engineering that takes everything into consideration and leaves nothing out. (See my review of the Vandersteen 2Wq subwoofer on the Vandersteen web site for even more information.)

Bass Crossover

A completely transparent, battery-biased, passive high-pass filter is inserted before the main amplifier. It creates a first-order (6dB/octave) bass roll-off starting at 100Hz (-3dB). The main amplifier is relieved of the task of delivering high current at low frequencies and deep bass frequencies are removed from the main speaker (not the subwoofer).

Intermodulation distortion is greatly reduced and dynamic range is greatly increased because the main amplifier no longer has to strain to produce high energy at low frequencies and the drivers in the main speaker can produce mid-bass and midrange frequencies undistorted by the large excursions required for bass.

This results in better sound. The main amplifier seems to be more powerful and the midrange seems clearer and effortless. Bass frequencies are delivered with control and authority by a true subwoofer driven by amplifiers designed specifically for this purpose. Deep bass capability improves impact, transient response and the ability to follow the rhythm and pace of music, and helps to expand the sound of the performance space.

The bass amplifier(s) samples the output signal from the main amplifier and corrects for roll-off created by the high-pass filter and the response characteristics of the bass driver in the enclosure. This compensation results in flat bass response that can be further tailored with the built-in 11-band equalizer to suit the speaker's position and the position of the listener, correcting for additions or subtractions contributed by the room.

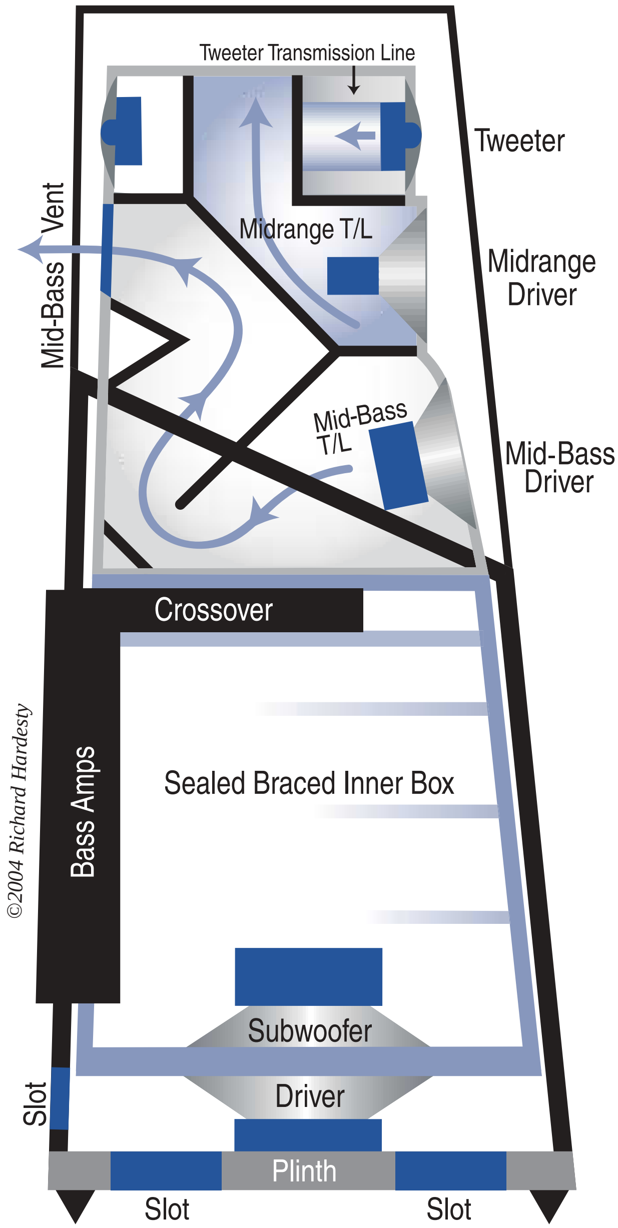
Feed-forward error correction and a unique "preview circuit" are included in the subwoofer system. The preview circuit analyzes the incoming signal and modifies the output as necessary, preventing electrical or mechanical misbehavior.

Because the bass amplifier(s) samples the output of the main amplifier, it passes along the sound and propagation speed of the amplifier that drives the main speaker system providing an ideal integration of subwoofer and main speaker. No other crossover method can provide this kind of seamless blending and transparency.

There are internal high- and low-pass filters that limit the subwoofer response to frequencies between about 7Hz and 100Hz. In addition to the eleven bass equalization controls, there are easily accessible controls to vary bass contour and overall bass level. These additional controls allow the speakers to be positioned where they image best and permit the bass to be tailored precisely to that location within the room and to the tastes of the listener.

Bass Enclosure

The unique Vandersteen push-pull subwoofer driver and balanced, bridged amplifier assembly are enclosed in a heavily braced, constrained-layer enclosure shaped like a trapezoid and mounted on a plinth made from a



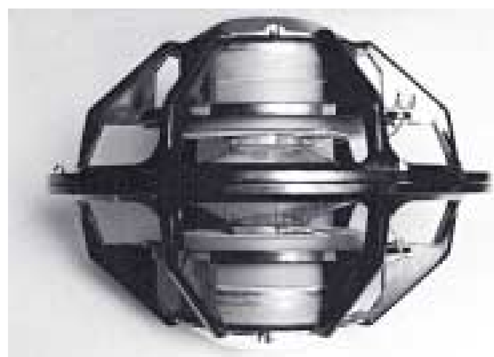
high-pressure epoxy laminate. This bass section represents the bottom two-thirds of the speaker and is finished like fine furniture. (We'll consider the upper module, which is concealed behind the removable grille, separately.)

The main bass structure is an enclosure within an enclosure. The internal structure is heavily braced by a series of cross-members with varied shapes. Resonances are randomized and dispersed and all are isolated from the external structure with a layer of adhesive that remains semi-viscous—the constrained layer which separates the two structures. The visible outer enclosure is acoustically decoupled from the inner structure that houses the bass driver. The modular bass amplifiers and

crossover network are also mounted in the lower portion of the speaker.

The plinth, beneath the enclosure within an enclosure, is made from an exotic high-pressure epoxy-laminate material that other manufacturers advertise as proprietary and magical. It is neither. It is, however, extremely stable and nonresonant. It's also very expensive and difficult to machine. This material is shaped with diamond cutting tools and is also used for the replaceable driver baffles on the front of the upper module.

Bass Driver



The 12-inch Vandersteen subwoofer driver is unlike anything else available. It has two complete motor systems with precision-formed magnet assemblies and copper faraday rings, which allow more than an inch of perfectly linear diaphragm excursion. These motors oppose each other and are supported by a rigid die-cast chassis, which resembles two driver baskets mounted face-to-face.

A single voice coil former, with a voice coil wound on each end, runs between the two motors and drives the diaphragm mechanism, which is comprised of two curvilinear aluminum cones sandwiched together with an exotic honeycomb material to form the most ridged diaphragm possible. This diaphragm is centered on the voice coil former and performs like a perfect piston within its pass band and well beyond.

Each voice coil is driven by its own amplifier and everything—mechanical and electrical—is mirror-imaged to cancel any non-linear distortion. One motor pushes while the other pulls. One voice coil encounters more iron while the other encounters less. One amplifier pushes while the other pulls.

Bass Amplifiers

There are four Class B bass amplifiers in each Model 5A speaker. They share a power-factor-compensated switching power supply but the amplifiers themselves have high-current linear circuits operating in a bridged configuration.

One bridged pair of amplifiers drives one of the opposing voice coils and another bridged pair drives the other voice coil. Amplifier nonlinearities are canceled along with driver nonlinearities in this completely balanced system.

Each amplifier uses a single pair of ultraheavy-duty bipolar output devices engineered to deliver very high current into the low impedance load of the subwoofer driver. The amplifiers, amplifier power supply, and subwoofer driver have been designed to work in complete harmony.

Because the driver has extremely long excursion capability and very high compliance and is mounted with the diaphragm in a horizontal position, the weight of the diaphragm and voice coil assembly would tend to allow the moving parts to be offset (sag) towards the floor. The amplifiers provide a levitating force to center the diaphragm and counteract this tendency so each bass note begins from dead center within the driver's range of excursion.

Upper Module Enclosure

The upper module houses the drivers that produce mid-bass and higher frequencies. While the upper module is an integral part of the complete speaker, it is constructed separately to enable it to resist resonances in the upper frequency ranges. This module is made from 22 layers of machined MDF. Each layer is .750" thick and the layers are laminated together using several adhesives to vary and disperse resonances.

The completed module is a solid billet of material that has been machined into the required shape, only stiffer and far less resonant. It positions the drive elements and provides the resistive transmission lines that load each driver. But that's not all.

A hidden cavity that contains a "secret sauce" of semi-viscous damping material surrounds the entire module. This produces an enclosure that has virtually no measurable resonant modes.

The drivers are mounted on removable plates made from the exotic high-pressure epoxy-laminate material used for the subwoofer plinth. This allows drivers to be replaced with new designs should they become available. These baffle plates provide an extremely rigid mounting surface for the drive elements and further isolate the drivers from the enclosure structure.

The front faces of these minimum-sized baffles are covered with a thick layer of felt. This eliminates any residual reflections that might emanate from the baffles, which are just big enough to support the drivers effectively. This is the Vandersteen "boxless" minimum-baffle design fully realized.

The upper module contains the drive elements that reproduce frequencies from 100Hz to 30kHz and beyond. This range is divided into three parts, which are handled by the mid-bass driver, the midrange driver, and the tweeter. We'll examine those next.

Mid-Bass System

The mid-bass driver is a 7-inch unit that has a rigid die-cast frame and a precision-formed magnet assembly with a copper

faraday ring to reduce magnetic distortion and allow maximum linear excursion. The surround is a unique co-injected fluoroelastimer rubber material molded into a distinctive shape that enhances performance. The diaphragm is a proprietary Vandersteen curvilinear poly/Kevlar® composite.

This driver is assembled by Scan-Speak using Vandersteen parts. Some manufacturers of more expensive speakers use the Scan-Speak catalog part with an off-the-shelf paper cone—as a midrange. You know the driver I'm talking about—it has razor cuts on the cone to break up resonances. This driver is actually a woofer and is too large to provide the speed and detail necessary for accurate reproduction of midrange frequencies. It's also too large to balance the polar pattern (dispersion) in the midrange.

Vandersteen uses this driver only as a mid-bass unit operating up to 600Hz and utilizes a special composite cone to ensure that it performs as a perfect piston within its pass band and well beyond. It is loaded by a resistive transmission line, which vents at the rear of the upper module. The purpose of this line is to eliminate in-band resonances and to flatten the impedance curve while preventing reflection of the rear wave back through the diaphragm, not to augment bass.

Diaphragm Materials

Vandersteen uses the best available materials for each driver diaphragm. The materials are not all the same because each one performs a different task. Each is optimized to perform over its range of frequencies while adding or subtracting nothing. The homogeneity of the sound of the complete speaker is a tribute to the success of this approach.

One further note: poly and polypropylene are not the same thing. Polypropylene is a milky-white plastic that was a popular diaphragm material twenty-five years ago. Poly is a new moniker that describes a blend of plastic materials loaded with various minerals for damping. Let me inject a bit of history here.

The British speaker manufacturer, KEF, pioneered plastic diaphragms with the introduction of polystyrene cones. Polystyrene was light and stiff and easily produced but had little internal damping. KEF added layers of PVA (polyvinyl acetate) front and rear to damp resonances and called the diaphragm material Bextrene. Damped polystyrene sounded pretty good but was no longer light and fast, resulting in speakers with very low sensitivity. Polypropylene materials were tried next because they provided high internal damping with less weight. Internal damping was probably excessive because polypropylene drivers sounded dead and lifeless (to me).

"Contrary to what you might have heard, Vandersteen hasn't used polypropylene in decades."

Materials have come a long way. Today manufacturers can blend plastics and add damping materials as needed to produce diaphragms with nearly ideal characteristics. Diaphragms can be light to increase sensitivity and reduce rise-time, stiff to perform in a more linear, piston-like manner over a wider range of frequencies, and highly resistant to resonances due to high internal damping. Contrary to what you might have heard, Vandersteen hasn't used polypropylene in decades.

The diaphragm in the Model 5A subwoofer is made from aluminum and honeycomb. The cone in the Model 5A mid-bass driver is made from Kevlar® with a laminated, mineral-filled poly skin. The self damping diaphragm in the Model 5A midrange driver is made from three different plastic fibers, which are woven together. The diaphragm in the Model 5A tweeter is made from an exclusive metal alloy that has been anodized to produce what is advertised as ceramic coated, and critically damped to eliminate resonance.

Midrange System

The unique 4.5-inch linear-surround midrange driver in the Model 5A is protected by worldwide patents and features several unusual design features. It has a "reflection-free" aerodynamic frame and magnet assembly with a copper faraday ring and small diameter alnico magnet, and features a diaphragm made from a proprietary woven material. It has a "linear surround" that can't go out-of-phase with the diaphragm as many surround rolls do.

Planar speakers sound very "open" partly because there is no reflective surface directly behind the diaphragm. Most midrange dynamic drivers have a large ceramic magnet, and the chassis to support it, positioned directly behind the diaphragm. The magnet and chassis can reflect the signal back through the diaphragm with only a slight time delay, smearing transients over this period of time. The Vandersteen reflection-free chassis has a small diameter magnet made from powerful alnico (aluminum, nickel and cobalt alloy) material. There is little behind the diaphragm to reflect midrange frequencies back to the listener, delayed in time. Reflections are minimized in other significant ways as well.

The Vandersteen midrange driver doesn't have a conventional surround roll that might reflect small vibrations back into the diaphragm material from the surround or produce distortion due to nonlinear behavior (flapping unrelated to cone movement or even out-of-phase with cone movement). Instead the diaphragm is terminated in a flat, nonreflective ring that acts as a suspension member allowing diaphragm movement while dissipating energy.

The diaphragm is made from a newly developed material that is comprised of three different plastic filaments, which are woven together to provide a highly damped and extremely rigid cone. This driver has flat frequency response to more than an octave above the crossover point to the tweeter.

The back of the midrange driver is loaded by a resistive transmission line designed to control driver resonance and impedance while absorbing rather than reflecting rear wave energy. This line is closed and rear wave energy is completed dissipated.

Each midrange driver and its crossover components are hand-adjusted in an anechoic chamber to produce linear frequency response (±1.5dB) and an exact acoustic match (±0.1dB) for its mate in the other speaker.

Tweeter System

A dual-chamber tweeter is one with a hole through the pole piece that opens into a rear chamber that can be adjusted to counteract diaphragm resonances. The first chamber is the volume of air between the diaphragm (dome) and the pole piece, and the second chamber is the volume of air in the hole and the cup at the back of the pole piece.

The Vandersteen tweeter has a proprietary resistive transmission line configuration that is inherently nonresonant and the diaphragm is meticulously hand-damped to eliminate the so-called "oil can" resonance that occurs at the first break-up frequency.

The diaphragms in all soft dome tweeters, including the fashionable ring radiators, start to break up (behave in a chaotic manner unrelated to voice coil movement) at about 10kHz. Stiffer metal alloys can be engineered to perform in a piston-like manner to a range beyond audibility. Eventually they too reach resonance and start to break up. Chaotic behavior ensues. At this frequency, usually 25-28kHz, they will exhibit a large rise in output (often 15dB or more) and a commensurate increase in impedance.

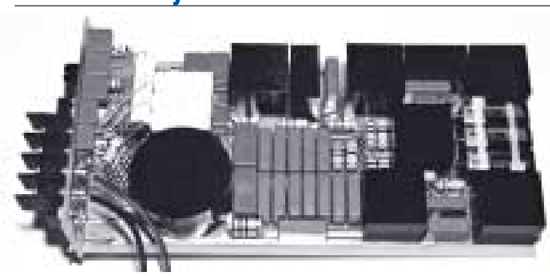
This ultrasonic resonance is ignored by many manufacturers because it's above the supposed range of human hearing.

John Atkinson (*Stereophile*) has said that only bats can hear it. I disagree. Empirical evidence gathered through listening tests shows that we can hear the results of ultrasonic sounds even if we can't perceive pure tones at those frequencies.

Vandersteen eliminates rather than ignores the resonance by employing an exceptional anodized metal alloy diaphragm and critically damping this diaphragm with careful hand-adjustment in an anechoic chamber. This results in specified frequency response to 30kHz (±1.5dB) and substantial output, albeit at reduced levels, to well above this frequency.

Each tweeter and its crossover components are hand-adjusted in an anechoic chamber to produce linear frequency response (±1.5dB) and an exact acoustic match (±0.1dB) for its mate in the other speaker.

Crossover System



First-order acoustic transitions and compensation for driver frequency response anomalies and impedance variations are accomplished by a battery-biased, balanced crossover network employing the finest parts available.

The Model 5A crossover uses metal film resistors with copper leads, premium film capacitors from Wima and InfiniCaps® with batteries to keep the dielectrics charged for optimum performance, and solid-core pure silver wire. Caps and hand-wound inductors are encapsulated to minimize microphonic effects.

The first-order crossovers look complicated but all compensation components are in shunt, not in the signal path. The signal follows the shortest and simplest path possible.

The crossover is completely modular and can be easily replaced or repaired. Each crossover network is hand-tweaked in an anechoic chamber to perform as specified with the exact drivers used in that matched pair of speakers. Left and right speakers are adjusted to produce frequency response from 20Hz to 30kHz with a maximum deviation of ±1.5dB. Left and right speakers are adjusted for output that matches within ±0.1dB across the entire spectrum.

You can spend a lot more but you can't buy another product with this level of engineering. So how does it sound?

Sound

I've spent a lot of time explaining the engineering features and the performance potential of the Vandersteen Model 5A speaker system. Let me assure readers that it all works just like it's supposed to. This is the most sophisticated design that I've ever examined and it provides the best sound I've ever heard—by a substantial margin.

The Model 5A's vertical array of drivers can provide a time- and phase-accurate simulation of a point-source over a limited vertical range (adjustable with spike shims) at a distance of 9 feet or more from the speakers. Listeners above or below a normal seated position, or closer than 9 feet from the speakers, will still hear good sound that won't be quite as phase coherent. A side benefit to this limited vertical dispersion is reduced floor and ceiling bounce, which means the room will typically have less negative impact on the sound of the speakers.

The model 5As have a balanced and uniform horizontal dispersion pattern, which means that they will be less sensitive to their position relative to the side walls. A three-dimensional image will require that they be positioned well forward of the front wall, a requisite that applies to virtually all speakers. Positioning the speakers a reasonable distance from the side walls—2 feet or more—will improve image focus.

> ***"…instruments and vocals float in space with a holographic, three-dimensional reality that can only be described as palpable."***

And focus they can! The near-complete absence of resonances and reflections provides a clear window to the performance. There is absolutely no sense of the speakers' locations and instruments and vocals float in space with a holographic, three-dimensional reality that can only be described as palpable. It feels like you can reach out and touch the performers. Each one has a precise and specific position on the soundstage.

Speakers with steep filters can deliver a good "mono-in-the-middle" image but instruments and voices towards the sides of the soundstage tend to pull forward towards the speakers. Time- and phase-accurate speakers can deliver images with depth and dimension across the entire stage. The Model 5As do this in spades producing the most convincing images I've heard.

The sound is remarkably smooth and free of distortion, yet incredibly detailed and resolved. Bandwidth is outstanding. No passive speaker system can rival the range of the Model 5As and none sounds as homogeneous.

In my room the system measures flat to below the range of my test instruments and I can hear low frequency information that is simply thrilling. Low bass and mid-bass are delivered with detail and authority that is unrivaled. I have reviewed more than 60 subwoofers and have never heard bass that can equal what the Model 5As can deliver. No full-range speaker even comes close.

"…this is the best speaker system available regardless of price."

Bass is not simply low and loud—it is finely detailed and textured. This speaker system allows the listener to hear more of what's going on in the music—all the way down to the deepest audible frequencies. There is an absolutely seamless transition from the lowest frequencies, which are produced by the subwoofer, through the mid-bass region where most musical fundamentals and vocals are delivered by the direct radiating mid-bass driver. If you have never heard a properly set-up pair of Vandersteen Model 5As, you have only heard a vague semblance of what's on the recording.

The midrange is mercilessly revealing. Does that mean that Model 5As can only be used with bleeding-edge associated components? Not necessarily. This speaker system is a relatively easy load to drive but it will reveal everything about the system components that drive it and the recording that stores the music.

Components with sins of omission are preferable because flaws that include additive colorations will be clearly exposed. Poor recordings are well tolerated by the Model 5A speakers but production flaws can be easily identified. When the best recordings are played through the finest components the results can be magical.

It's difficult to criticize any aspect of the high frequency response. It's clean, clear, extended and completely free from zing and exaggeration. There is virtually no sound from the enclosure providing an openness and image focus that is unrivaled in my experience. The tweeter doesn't stand out like a ribbon or provide an artificial sense of detail like some inverted domes. It blends perfectly with the rest of the speaker providing a completely integrated sound.

There is a clarity and freedom from strain that allows dynamic contrasts I didn't know were possible from an audio system. This is mostly the result of reduced distortion, but freedom from resonances and reflections probably contributes to this relaxed sense of ease. I have a preamp that can deliver more than 30 volts and mono amplifiers rated at 400 watts each, which can't hurt. The system is dead quiet and can play loud enough to make your ears bleed. Dynamic range is startling.

Proponents of speakers with steep filters claim that first-order speakers won't play loud and suffer from driver strain. Vandersteen has overcome these supposed deficiencies with a powered bass system and a 4-way design that uses exceptional drivers. Think your horns have greater dynamic range and less distortion? Come visit my house.

I could continue with additional superlatives but an audition would be better than another thousand words. I think this is the best speaker system available regardless of price. I put my money where my mouth is and bought a pair for myself. Although I've probably owned more speakers than most people, I suspect that this is my last pair. APJ

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An Interview with Pat McGinty

by Richard Hardesty

When Dunlavy Audio closed, the small assemblage of established manufacturers making time- and phase-accurate loudspeakers shrank by a substantial margin. Pat McGinty, speaker designer at Meadowlark Audio, along with Jim Thiel and Richard Vandersteen, of Thiel Audio and Vandersteen Audio, respectively, are in an elite group that continues to make speakers which create an acoustic output that replicates the electrical input signal. I'm convinced this matters, and in this interview we'll find out what Pat thinks.

> ***"…combined harmonics provide the timbre or color that identifies the instrument."***

Each musical note has a fundamental frequency and each instrument or voice includes a unique set of harmonics or overtones. The combined harmonics provide the timbre or color that identifies the instrument. Experience has shown me that these multiple frequencies need to be reproduced with the proper phase relationship in order to provide tonal accuracy and correct rhythm and pace (called tempo in music). Many speaker manufacturers, and publications supported by their advertising, would have you believe otherwise: that time and phase relationships are inaudible and only frequency response matters. Let's get Mr. McGinty's opinion.

Pat, tell us something about your background. Why did you choose a career in the audio industry?

Looking back, I realize that I began at a very early age to acquire the skills that would later coalesce into a career in audio. I have been very fortunate to have a few excellent teachers starting with my dad, who had been working on the frontier of radar technology with the Navy during WWII and returned to become an instructor at the Naval Academy for a time, then entered the civilian sector and dug right into another emerging technology: television. He took part in RCA's successful crash program to be first with color TV and to establish the FCC standard, vanquishing Zenith. So he really, really knew his stuff.

You sort of inherited an interest in the science that is the basis of our industry?

Those were the times of Dr. Spock so my dad had the clever idea that a child could learn electronics at the same time he acquired language. I did just that, having no way of knowing it wasn't exactly normal until I entered kindergarten with a very good understanding of DC electronics and discovered that I was completely alone. My education expanded from there and a more caring and masterful instructor has never taught!

We journeyed through much of classical physics too. I still recall being fascinated by the details of the Michelson Morely experiment as a bedtime story! I've made it a point to maintain a ready knowledge of the Classical Physics as well as to stay conversant in the Einsteinian and Quantum branches. One look at my darn library would really give me away!

You also have amazing skills in woodworking. How did that come about?

During my teen years I found a strong love for the art of woodworking, built several boats from my own plans and did some other pretty tricky and showy projects. Later in life I had the remarkably good luck to have studied cabinetmaking under Mr. Ian Kirby, the undisputed Planetary Grand Master of fine hand cabinetmaking, during his tenure at Palomar College in California. I learned from Ian the subtleties of method, design, construction and philosophy that I was unable to teach myself.

I assume that an interest in music was intermingled with these mechanical and scientific endeavors.

Let me back up a few decades. Music, particularly classical music, was an ever present part of my upbringing. I lived close enough to New York City to make frequent trips to enjoy Leonard Bernstein and the New York Philharmonic and to attend concerts at Carnegie Hall. I was just the right age to enjoy the amazingly beautiful NY rock scene during the years from 1968 to 1971. I was a regular at the Fillmore East during its heyday.

Anyway, at a point about twenty years ago, those three key factors—a comprehension of electronics and the sciences, an ability to create the right cabinetwork, and a love of music—somehow precipitated into a fascination with speaker building on the hobbyist level.

How did you get beyond the "hobbyist" stage?

For a few years I hit the books pretty darn hard to firm my grip on the subjects of wave mechanics, magnetics, materials, resonance and filter design all the while building a ton of crazy project speakers. A constant stream of experiments overflowed from my garage to the rest of the small home that I shared with my two daughters and a wife, who, upon reflection, was more indulgent of my obsessions than made any sense at all. Worse

yet, large sums of money went into the purchase of a rather extensive test bench that was to eventually allow me to pry out the secrets of speaker design and performance.

I do regret having not kept a journal with photos that, today, would make for an interesting and very funny story. In a world where many speaker designers seem to wrap themselves in the purple robes of genius, I'm not ashamed to tell you that I built somewhere between 100 and 200 pairs of speakers before they started to actually sound really good and before the ideas that were to form my design philosophy finally began to emerge.

During that time I took on numerous custom jobs that were ascendingly complex and ambitious; some knockoffs, some commercial jobs. So, other people's money funded a period of intense experimentation and discovery that began, oddly enough, with the idea that the "way to go" was fourth-order closed boxes!

But that position changed as you progressed towards becoming a commercial enterprise?

As my understanding and experience grew it became clear that Time Coherence was fundamental and that transmission line bass systems worked quite well. I began to look at all of the aspects of accurate waveform reproduction and incorporate them into projects that began to look more and more like production speakers.

Early in 1994 I began to think that my latest project might be as good as the speakers being offered by the "big boys." So I loaded the pair into the truck and headed off to offer them to a prominent retailer in LA. He bought them on the spot! I remember driving home and thinking, "Darn! Now I'll have to remember exactly what I did!" Anyway, he told his buddies, who told their buddies and business just took off like a rocket. The original Kestrel was a floor-standing, time coherent, transmission line that sold for $995/pr. I spent the next four or five years straining to make enough of them and, believe me, the business challenges were just as difficult as the engineering challenges.

"As my understanding and experience grew it became clear that Time Coherence was fundamental"

In retrospect we had done the impossible: we entered the national market with a single speaker—not a whole product line—with hardly any money, no marketing and no plan. The endearing qualities of the Kestrel were enough all by themselves.

It's far more difficult to make time- and phase-accurate speak-

ers than it is to make speakers that have flat frequency response only. Why is time domain performance important?

You're right. I've designed at least as many steep filtered speakers as coherent ones and it certainly is true that coherent designs are both more challenging and much more time consuming. Any competent designer can take a set of qualified drivers, put 'em in an enclosure and force a flat response with a complex filter. It's easy with the help of today's filter design software and a good pair of ears. You can finish up in a few hours, days maybe.

On the other hand, the qualifications for drivers in a coherent speaker are much tougher. The work must begin with establishing correct timing and the filter work is much, much more

tedious and drawn out, easily taking many days or weeks. Sometimes you can't reach your goal and have to go back to the beginning. That doesn't happen on steep filtered designs.

So, aside from the issues of increased cabinet costs and the necessity of an increased driver budget, you can see why any designer employed by one of the bigger speaker companies would not risk a venture into the realm of coherence; a few weeks of filter work would never fly!

Really, achieving a flat response is no trick at all. Take that box of drivers and you can come up with any number of flat solutions, most of which will sound bad. What does that tell you? It tells you that a flat response is not an appropriate design goal. Instead, a flat response is the RESULT of good design, which means a bunch of other things—paramount among them is, for us, time coherence.

Ultimately, coherence will be recognized as a minimum requirement in speakers just as it is now for microphones, cables, cartridges, players, preamps and amps. But for the moment, it is the province of a very few, highly dedicated speaker companies with clear design philosophies.

"Ultimately, coherence will be recognized as a minimum requirement in speakers…"

The whole issue of coherent vs. incoherent really boils down to this: Do you want to reproduce the waveform? Yes or no?

If you believe that the whole idea of audio is to come as close as possible to reproducing the sound heard by the microphone at the original performance, there's not much more to think about.

So you feel, as I do, that speakers must be time coherent to be truly accurate?

Let's be clear about what's happening in the audio chain beginning with the performer who creates sound, which arrives at the microphone diaphragm in the same form in which it arrives at your eardrum: a two-dimensional function consisting simply of pressure and time, where it is transduced into a simple voltage/time function for storage and later reproduction.

It is interesting to note that the inventor of sound recording figured out the pressure/time thing over one hundred years ago! Thomas Edison realized that he could transduce the pressure/time function at a diaphragm at the bottom of a big horn to a displacement/time function at a pointy stylus and that, if he moved something impressionable like wax past that stylus at a steady speed, he could store the sound as a displacement/length function and later reverse the process! My hero!

Both your phonograph and your CD player do excellent jobs of spitting out the waveform that was impressed upon them, which is then amplified—a process that merely increases the magnitude of the voltage in the voltage/time thing, after which the signal is sent to the speakers where, in most cases, the time part of our simple two-dimensional signal is badly mangled.

Why does this happen? The first aspect is really just an artifact of the way woofers and tweeters are physically made: their ordinary shape and mounting scheme mean that simply mounting them on a vertical baffle board actually causes the tweeter's acoustic output to arrive at the listener a discrete measure of time earlier than that of the woofer.

The second aspect is that the filters used to send treble to the tweeter and bass to the woofer are bound by their fundamental nature to make a mess of the signal's timing. This is an interesting thing that is well understood and descends from the basic physical laws of the universe. Think about the relationship between velocity and acceleration. Any change in velocity causes a very predictable acceleration. Acceleration is the derivative of velocity. We employ filter circuits in a speaker to achieve attenuation of the signal with either ascending or descending frequency. Therein lie the Devil's details: as when we change velocity quickly we cause big acceleration, when we attenuate quickly we inescapably cause big timing errors because they are tied to attenuation as its derivative.

"…all that can be achieved in a filter circuit is attenuation and delay."

Keep in mind that all timing errors are delays—because, as we know, time only goes in one direction! So, once an error occurs, it cannot be undone; all that can be achieved in a filter circuit is attenuation and delay.

Some people believe that simply staggering the drivers can correct for timing errors.

The situation is a bit more intricate than just that. The errors do not occur in discrete measures of time, otherwise we could easily "fix" the mess by merely moving the tweeter, whose signal might have been delayed by some discrete amount, forward towards the listener in order to make up for the delay caused by the filter. The problem is that the delays come in bits called "phase," which is an amount of time that changes with wave-

length. Worse, because the pressure/time thing naturally swings between compression and rarefaction, just the wrong amount of phase applied to compression will turn it inside out into rarefaction. So, you have the situation where not only is the relative timing between bits and pieces of the waveform rearranged, some bits of the waveform are inverted with respect to others. You can easily imagine that a signal passed through such a device would bear little resemblance to the original signal.

It always strikes me as ironic that a great number of audiophiles pay intense, careful and loving attention to the tiniest aspects of their system's design and performance—right down to worrying about the dielectric in their interconnects—but are oblivious to the severe timing distortions occurring in their speakers!

Some people believe that you can't hear phase errors but we know better, don't we?

The question remains: Why is it important? Some argue that human hearing is oblivious to those timing errors and I don't doubt that for some individuals that is entirely true. On the other hand, I hear constantly from experienced listeners who just cannot tolerate incoherence. In the middle are the majority of audiophiles who, I think, have just not yet recognized the value of accurate timing in terms of their own perceptions.

Human perception of sound is complex. Our hearing system evolved, as it did in other species, as a means of survival. Your two ears present a subconscious part of your mind with two discrete signals that contain the pressure/time information impressed upon your eardrums for processing into a perception that is then presented to your consciousness. Both the relative amplitude and relative timing information are employed to get the job done.

You are quite capable of very fine discrimination in the time domain. Imagine you are in the woods at night and you hear a twig break nearby. What you are actually hearing is the original twig break followed very shortly thereafter by numerous reflections from nearby trees and the ground. If your conscious mind were presented with all of that data to process, you might soon be something's dinner. But your mind can sort things out rather nicely and, it is important to note, effortlessly based on very tiny timing clues.

"...when the data conflict, perception suffers."

In stereo we are doing our best to create a perception of a soundstage that is clearly not true, but an illusion. Your mind sorts out the pressure/time signals and forms a perception just as it does in the "real world." But when the data conflict, perception suffers. Take an extreme case—wire one speaker out-of-phase. Now, you're getting good amplitude info, but conflicting timing info. So, that part of your brain that is in charge of sorting things out just cannot get the job done and no perception of a soundstage or focused image forms at all.

Now there's a gross timing error that few would argue is inaudible.

When you listen to time incoherent speakers the situation is somewhat better, but what happens is the processing system in your mind has to work a whole lot harder to form a perception.

That, I believe, is the primary cause of listener fatigue. Personally, I experience a lack of emotional connection to the music and an increased awareness that what I'm hearing is not real. It's a thing called musicality, or the lack of musicality.

The most often reported complaint about incoherent speakers is that they seem to fail to engage the listener over longer periods of time. That explains why guys have rigs that seem to be doing everything right, but fail to hold their attention and interest in the music. They sit down to listen, their mind wanders and the next thing they know they're in the kitchen making a sandwich!

Too, perception of sound is a learned skill. For instance, you can readily identify your wife's voice in a crowd but I cannot. But that situation would change easily if I got to know her. Once a listener cues in to the sonic problems that are caused by timing errors, they will bother him forever, or until the problem is corrected. For example, a while ago I pointed out to a fellow, who was immensely proud of his rig, that there was a very tiny, unnatural sound that occurred at the moment of hammer fall on piano; kind of a little, springy "oink" instead of the flat "clack" that you'd expect if you knew how a piano really sounds. It was, of course, the timing screw-up generated by his fourth-order filters. He'd never noticed it before but was, from that moment of recognition, driven crazy by it until he finally dumped the speakers.

Similar perception problems occur most often on transient and asymmetrical waveforms: percussion, strings and horns. Perhaps the more familiar the sound is to you the more critical you are of timing accuracy. I suspect that's why I personally find coherence to be the key to believability of reproduction of the human voice.

Certainly, we've all had the experience of buying audio gear that sounded good to us the day we bought it, only to like it less and less as time passed. That scenario seems to play out much more frequently with speakers and I suspect that poor timing is the reason why.

In the end, incoherence is wrong and takes us a step away from the objective of realistic sound reproduction and the goal of complete believability. When you're trying to reproduce a pressure/time function keeping the timing correct is obviously not just a good idea but a necessity.

Proponents of steep crossover slopes will cite increased intermodulation distortion and reduced dynamic headroom as negative aspects of first-order crossovers. How would you respond to these criticisms?

Yes, there certainly are things that you can attain by trading away timing fidelity. But the real question is: Are the things I'm trading towards worth more than what I'm trading away? You constantly hear of the advantages claimed by the steep filter crowd. If they really think that giving up proper timing for some other objective is a good idea, then they should do it. But, invariably, a good explanation of why those trades were made is lacking.

To answer your question, if your number one objective is to get maximum peak SPL from a given set of drivers, then yes, by all means, apply a steep filter! But admit that you've just made the deliberate decision to trade away time coherence for big output!

In practice, it doesn't matter because if you're building a coherent speaker, you already know that you're going to have to use "more driver" to achieve the same peak SPL as an equivalent steep filtered design. So, there's no problem. You spend a few bucks more on drivers, the same few bucks that the steep filter guy spent on all those caps and coils!

IM is a different situation. If you're trying to set up a time coherent speaker and it exhibits audible IM, your next choice is certainly not to apply a steep filter. It's hard to imagine that you'd deliberately trade one form of distortion for another. Your choice is very simply to reduce the IM to a level that is below the audibility threshold. Happily IM is both easy to measure and easy to perceive. I experience it as subtle clicking sounds that seem to originate right in my ear canal.

"It's hard to imagine that you'd deliberately trade one form of distortion for another."

But, frankly, it's a nonissue. Audible IM or even high levels of measured IM are just not part of the everyday problems we face when putting together coherent speakers.

The truth is that most vexing problems of speaker performance can be completely worked out within the context of time coherence and that, most often, the choices that are made by steep filter proponents are made for some combination of economy and expedience. The criticisms of time coherence are mostly defensive ones made by sellers of incoherent products. As one of my customers so aptly put it, when he realized that he had allowed another speaker maker to fill his head with doubt, "He sure put a monster under my bed!" referring to a scary fiction from childhood.

There is no doubt that, in speaker design, choice has its price. But I say: make your design choices for good reasons that you can explain and that you can stand behind. In our case time coherence tops the list.

Besides time domain accuracy what, in your opinion, are the most important requirements for good sound from loudspeakers?

By now you should not be surprised when I tell you that most of the important performance criteria fall under the general heading of waveform fidelity. Time coherence plays a big part in seeing to it that the speaker faithfully reproduces the waveform, but there are other key ideas as well.

On the frontier of our art is the problem of dynamic linearity. In a case of ideal dynamic linearity the magnitude of the acoustic output increases exactly as the magnitude of the input signal, at all frequencies within the band. Every time you double the input, you should see a doubling of the output. When you see less than that, the waveform is not being accurately reproduced, but rather is being "compressed." This effect is one of the main reasons why stereo sounds like stereo instead of sounding like the real thing.

"This effect is one of the main reasons why stereo sounds like stereo instead of sounding like the real thing."

A speaker can have flat amplitude response, correct timing, excellent detail resolution, wide bandwidth, freedom from ringing and other artifacts and still sound completely unrealistic due to the very common limitations on dynamic linearity. Our top-of-the-line speaker cuts some new ground in this regard and the perception of explosive ease across the entire band is really quite thrilling. Good dynamic linearity does not come cheap, but it is certainly worth the price.

Another key issue is bass ringing, or rise- and settle-times in the bass. In a market that is very much driven by specs, one of the prime measurements by which customers compare speakers is F3 [the low frequency limit of the system or –3dB point—Ed.], which is the customary indicator of bass extension.
Bass systems are in fact resonant systems. They store energy and release it later—just like a bell and, so, they involve another of audio's many ugly trade-offs. All other things held constant, the price that is paid for a lower F3 is increased ringing.
When we look at bass systems on the test bench we excite them

with a transient waveform and watch for the amount of time it takes for them to reach full amplitude. About the worst I've seen was another maker's subwoofer that, at 20Hz, took 10 cycles to achieve full amplitude, as it stored energy. Then, when the signal was removed, it took another 10 cycles to quiet down as it released it. Now, 10 cycles at 20Hz is a half-second! Now that's ringing! The funny thing is that the speaker was definitely producing all of the bass its maker claimed it should, but just way too late. But, hey, it had a great—and completely honest—spec! The problem, of course, is that the system was definitely not reproducing the waveform, not even close.

We also look very carefully at our systems for rise- and settle-times across the entire band because that is a crucial indicator of waveform fidelity.

Another, more obvious area where a speaker can make noises that are not part of the input waveform is cabinet radiation. When a cabinet panel is storing energy and releasing it later, it certainly bears no resemblance to the input signal. Stiffness and mass combined with the right strategies for damping are the keys to licking that problem. For us, the numerous internal partitions that we need to form the transmission line do double duty as cabinet stabilizers. Today, accelerometers are cheap and testing methods are easy, so there's really no excuse for a noisy cabinet.
Internal reflections from within the cabinet and from the drivers themselves can also present troublesome deviations from waveform fidelity. Driver suspensions can be pretty noisy and very many tweeters make prominent reflections from their pole pieces and cavities.

So, you get the idea. We approach the problem from the perspective of whether or not the speaker is accurately reproducing the waveform and, of course, no other noises. We know that our job is to recreate sound in your room that is identical to the sound recorded by the microphone. A simple idea, I think.

You're fond of transmission lines for bass loading. Can you tell us why?

A properly designed and executed transmission line does a remarkably good job in the bass unlike an ordinary vented alignment, which is a resonant system consisting of the driver, the elasticity of the trapped air in the box, which acts like a spring, and the mass of the air in the port, which lowers the fundamental resonance frequency.

The vented system really works by ringing, hopefully not too much. In a TL we introduce the principle of a resistive load that is fundamentally more damped than the simple spring/mass idea of a vented box. The trade we are making is in favor of lower F3s with less ringing so our bass systems track the input waveform much more faithfully. The price we pay is that TLs are bears to design and much more difficult and expensive to manufacture.

I've been working with TLs for many years and still approach each new design with the expectation that I'll need to go through numerous prototypes to get it right. The process is an iterative one of building and testing, first by looking at the impedance curves alongside rise- and settle-times, then by finding the limits of dynamic linearity and finally by listening. My crew usually takes the chance to poke fun at the pile of castoffs that often builds up in the corner of my lab as I work to get it exactly right. We call it "dumpster fodder!"

It's hard to imagine a designer who works for one of the biggies going through that kind of process anywhere near his boss. Vented boxes are much safer, but they just don't work quite as well.

Are special manufacturing techniques required to make high-performance speakers?

Sure. Early in our history as a company we were limited in what we could do to designs that could be accomplished in an ordinary cabinet shop. Our first line of speakers reflected the limitations imposed by table saws and hand-held routers.

But in 2001 while much of the industry began outsourcing to China, we built a wonderful new factory in the nether regions of northern New York with just the right tooling to build our next generation product. Today, I'm pretty much free to build whatever I can imagine.

But one of the beautiful things about our plant and our methods is the freedom we have on the business side of the equation. Most methods, particularly the outsourcing or subcontracting methods of making cabinets, rely on enormous batches to work properly. The problem there is that, for each new model you wish to bring to market, and for each finish color you wish to offer, you need tons of inventory. So the main obstacle to advancing new product is often money, not design. Here, we have pioneered a totally new, non-batch process that allows us to turn raw materials into finished speakers in about one week, thereby mimicking inventory without the horrible expense. So we are now free to expand our line without the usual financial drag. We can also do something that very few speaker makers can do, and that is to respond quickly to orders for custom finishes and variations. That niche business has proven very valuable to us and to our dealers and customers.

"Our first line of speakers reflected the limitations imposed by table saws and hand-held routers."

What can we expect to see in the future from Meadowlark Audio?

We are on a roll. In the past few months we have introduced the new American E Series of below $1K/pr. time coherent speakers, a new subwoofer and a few big, all-out, time coherent center channels.

At present I'm working on a completely new idea for a petite series of very exquisite speakers made from exotic materials and extremely esoteric parts. They'll be expensive, they'll look expensive and they'll sound expensive but I'm going to keep the details under wraps until we're ready to go.

Following that we have a wonderful idea for, of all things, an in-wall speaker. And there seems to be very strong demand for a $30K statement model but the ideas have yet to form about exactly what that will be like.

I think you'll see constant refinement of our products and I can promise you that we will continue to pursue improved technology of waveform fidelity and, too, we'll continue to make 'em right here in the states.

Thank you, Pat, for an informative discussion. I'm sure ***Journal*** *readers will learn a great deal from your explanations.* **APJ**

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Conclusion

While most audio components can deliver high fidelity performance—where what comes out is an accurate representation of what goes in—most speakers cannot. The vast majority of loudspeakers available today can't produce an acoustical signal that replicates the electrical signal delivered by the amplifier. Even though I'm confident about my position, the degree of importance of this fact is debatable.

High Fidelity?

I believe that you'll be happier, longer with high fidelity loudspeakers that accurately reproduce the recording. Others can effectively argue for other approaches. In the end, if you aren't pleased by hearing what's on the recording, then arguments about the accuracy of reproduction are moot. We're talking about home entertainment after all. The most important factor is that the listener be entertained and that can't be neatly quantified.

Maybe a certain coloration, though technically an inaccuracy, makes your listening experience more pleasurable. If that's the case, and it's particularly likely for those who listen mostly to a single musical genre, stand tall. It's your house and you should furnish it with things that make you happy.

I have no problem with people who choose speakers that produce sound that pleases them, even if that sound is unsatisfactory to me. The information offered here is my attempt to make sure they aren't being fooled. If you choose speakers that are demonstrably inaccurate, buy them because you like the sound they produce, not because you've been told that an exotic design changes the laws of physics or advances the science of audio reproduction.

If you choose speakers that cost a lot more than I think they should, buy them because they offer something you want—pride of ownership, prestige, great appearance—not because you think that a high price guarantees better engineering or higher quality.

Other Ways to Satisfaction

The distinctive sound of horn speakers reminds some listeners of what they heard from the sound reinforcement speakers used at live concerts or in the theater. This can trigger a favorable response in some people that provides exactly what they are looking for in reproduced music. Would I tell them they shouldn't like what they're hearing? Not on your life.

The sound of planar speakers reminds some people of the spaciousness of the concert hall and allows them to forget that they are listening to a reproduction rather than a live musical event. This may produce "goose bumps" or whatever the desired emotional response is, from recorded music. Should I denigrate their choice because my measurements show a difference between the input signal and the acoustic output of the speakers?

If you like what you hear, by all means buy it. Don't force yourself to live with something you don't like because I said it was more accurate. A home entertainment system should entertain you in your home. But be aware of what you're buying. Remember some simple facts that can prevent you from being conned and then spend your money any way you see fit.

Facts About Performance

This **Journal** explains why most speakers aren't capable of accurately reproducing the signal from the amplifier. You'll have to do some listening to determine whether accuracy is important to you.

Some products are inherently inaccurate yet provide some sonic characteristics that some people find appealing. Some products are actually designed to be inaccurate in order to provide a sound that is seductive to some people some of the time. Compare these products and come to your own conclusions.

Facts About Cost

Planar speakers are cheaper to manufacture than dynamic speakers. If they cost more, be sure they offer some other quality that you value more than measurable performance.

Augmented 2-way speakers don't have a real midrange driver and can't provide the detail and dispersion characteristics of 3-way (or more) speakers that do have real midrange drivers. Many of the most expensive designs available fall into this category (augmented 2-ways). Be aware that you'll be paying for something other than performance.

If a $30,000 speaker system provides you with prestige or pride of ownership, then it has justified its cost for you. Don't buy it because you assume it performs better because it probably doesn't. Don't buy it because you assume that it is made to higher standards of quality because it probably isn't. How can you tell? Read **Journal** 13.

What's Coming

You can't tell everything about how a speaker sounds by examining measurements but you can certainly tell if the speaker has the potential to accurately reproduce the signal. These measurements are readily available in *Stereophile Magazine* (and from other, less reliable sources) but most people have no idea how to obtain useful information from them. In **Journal** 13 I'll tell you what the measurements mean and explain how to interpret them.

Most speakers have compromised bass performance and I'll show you why and explain why most subwoofers only make the problem worse.

Journal 13 will include an interview with Richard Vandersteen and maybe another influential speaker designer. Readers can learn a great deal from interviews with the engineers who struggled with the design choices that resulted in the final product. APJ

Dispelling Common Myths

This **Journal** is all about speakers—specifically speakers that reproduce music. It is commonly believed that speaker crossover networks divide up the musical notes into various ranges and send each of these to the appropriate driver for reproduction. It's not that simple.

If, like most people, you assume that high notes emanate from a tweeter and low notes come from a woofer—think again. Music is far more complex than that and the notes occur at much lower frequencies than you might imagine. Each note is made up of a fundamental and the harmonic structure that adds timbre.

Examine the graphic at the end of this **Journal** and see the actual frequencies of musical notes. Then consider the following facts. The piano keyboard spans the range of the entire symphony orchestra. Only a few pipe organs, which are not conventional symphonic instruments, can play lower and no acoustic instruments can play higher. The highest "C" on the piano is the highest note there is and it's only 4,186Hz. The lowest note on the piano, an "A" at 27.5Hz, is 7 keys below the lowest note on a bass, an "E" at 41.2 Hz!

Bass is much more than a sound effect—it's the foundation, tonally and rhythmically, of all music. Nearly all melodies are confined to the single octaves above and below middle "C." The lower notes provide bass harmony and rhythm. The upper notes provide harmony, and harmonic overtones create timbre.

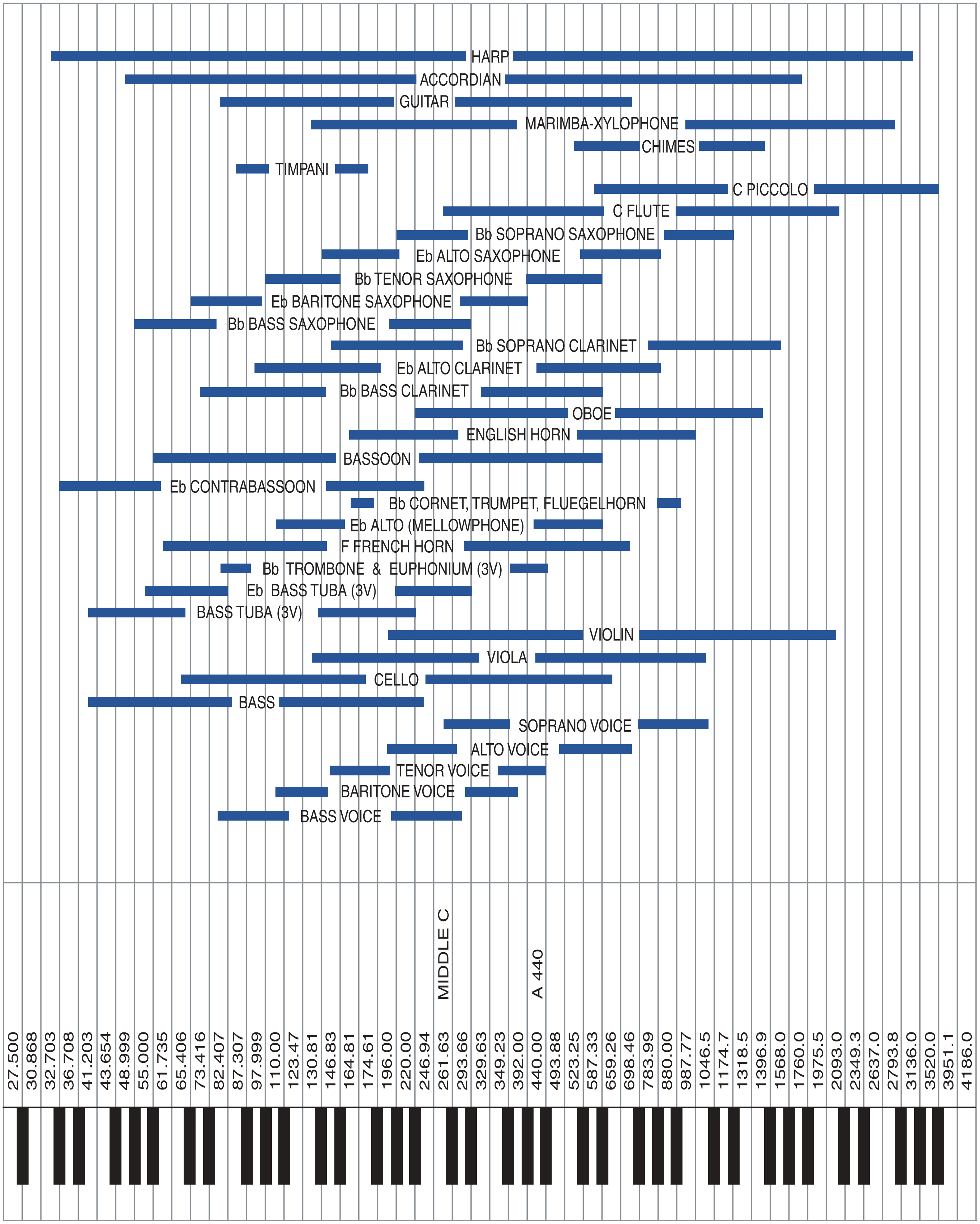
A bass voice can sing nearly two octaves below middle "C" and a soprano voice can reach two octaves above middle "C." The bass voice will dip into subwoofer territory at 80Hz or so, and that ultrahigh note from the soprano will still be only slightly above 1kHz, nowhere near the range of the tweeter in a typical 3-way speaker. Surprising huh?

The frequencies shown in the illustration are fundamental frequencies. You'll need an audio system with the capability of producing much higher frequencies (well above the range of human hearing) in order to accurately recreate timbre. Timbre consists of the fundamental note and the exact combination of overtones, which distinguishes one instrument or voice from another by adding unique musical characteristics (color).

You'll need an audio system that preserves the timing relationships between the fundamental and harmonics in order to accurately recreate timbre and a correct acoustic image.

The Whole Speaker

The entire speaker may be utilized to reproduce parts of one note. The fundamental that defines the note may emanate from the sub-bass or bass driver. The harmonics that provide timbre may emanate from the midrange and tweeter drivers at the same time. Don't let anybody tell you that timing (phase) doesn't matter. They're either selling speakers that can't do it right or they have not yet learned to hear this form of distortion. APJ


HARP
ACCORDIAN
GUITAR
MARIMBA-XYLOPHONE
CHIMES
TIMPANI
C PICCOLO
C FLUTE
Bb SOPRANO SAXOPHONE
Eb ALTO SAXOPHONE
Bb TENOR SAXOPHONE
Eb BARITONE SAXOPHONE
Bb BASS SAXOPHONE
Bb SOPRANO CLARINET
Eb ALTO CLARINET
Bb BASS CLARINET
OBOE
ENGLISH HORN
BASSOON
Eb CONTRABASSOON
Bb CORNET, TRUMPET, FLUEGELHORN
Eb ALTO (MELLOWPHONE)
F FRENCH HORN
Bb TROMBONE & EUPHONIUM (3V)
Eb BASS TUBA (3V)
BASS TUBA (3V)
VIOLIN
VIOLA
CELLO
BASS
SOPRANO VOICE
ALTO VOICE
TENOR VOICE
BARITONE VOICE
BASS VOICE
MIDDLE C
A 440
27.500
30.868
32.703
36.708
41.203
43.654
48.999
55.000
61.735
65.406
73.416
82.407
87.307
97.999
110.00
123.47
130.81
146.83
164.81
174.61
196.00
220.00
246.94
261.63
293.66
329.63
349.23
392.00
440.00
493.88
523.25
587.33
659.26
698.46
783.99
880.00
987.77
1046.5
1174.7
1318.5
1396.9
1568.0
1760.0
1975.5
2093.0
2349.3
2637.0
2793.8
3136.0
3520.0
3951.1
4186.0

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