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Speaker Measurements

Objective Gauges of Fidelity

*This **Journal** will explain how to interpret speaker measurements so you'll know what the test results mean and what they can tell you about a product's potential for accurate performance. These objective measurements can be very valuable to consumers because they show which products to avoid thereby narrowing the field of contenders and minimizing the number of components you'll have to audition.*

Objective measurements don't tell the entire story of course, and they raise the age-old questions about objective versus subjective product evaluation.

The Debate

A debate we encounter frequently in audio involves the validity of "objective" versus "subjective" evaluations. You'll notice that I have enclosed both terms in parentheses. That's because neither word has an exact definition when it comes to audio, and it's easy to become involved in a semantic argument that simply diverts attention from the real issue. Objective usually means an evaluation based on repeatable "scientific" tests. Subjective usually means an evaluation based solely on the emotional reaction of the reviewer. Ah, if life were only that simple.

In reality neither method can be employed alone in order to come to satisfactory conclusions about audio components. I'll offer examples in an attempt to explain why.

If you hate listening to a component or system, no group of measurements can magically make that experience enjoyable. On the other hand, if a component or system is demonstrably inaccurate you'll learn to hate it sooner or later (based on my experience) when you learn how to hear the flaw(s) that may initially have gone unnoticed.

I'm convinced that you have a far better chance for long-term satisfaction if you follow the high fidelity approach and choose from those components that accurately reproduce the signal according to accepted standards. You can still rely on your subjective responses as you choose from those components which are objectively accurate.

We live in a touchy-feely world and many will try to convince you that "if it sounds good, it is good." While there is an element of truth here, there are objective measurements to gauge fidelity and many will try to ignore this fact or downplay its significance. Why do they do this? Because in a world where no objective standards exist, anybody can be a "designer" and everybody is an "expert" or qualified "reviewer." Where would all these pundits be if they were forced to learn something about engineering?

Semantics

An argument about the meaning of the words can divert attention from the real issue. That issue is deciding which method of component evaluation produces the most satisfactory results.

What's objective? A conclusion based solely on "facts" perhaps? Whose facts? Which facts? Can we come to a truly objective conclusion based on only a few selected measurements (facts)? If so, which ones count? Among those that count, which are most important? Do measurements reflect all the facts we need to examine in order to come to a satisfactory conclusion?

And what's subjective? A position based purely on emotion? Is the emotional response of the examiner a faulty basis for conclusions? Isn't an audio system designed solely to produce an emotional response in the listener?

I can always identify my wife's voice and usually identify the brand of piano playing. Doesn't that make me a skilled listener who doesn't need measurements as a guide? On the other hand, couldn't I be fooled by products with colorations complementary to my wife's voice and pianos?

If a certain coloration reminds me of a certain kind of music some of the time, isn't that enough? However, if I like a certain coloration won't my listening be limited to a genre that benefits from that coloration? If I listen exclusively to that musical genre does it matter that measurements may show that the sound I enjoy is not accurate? If I hate listening to an audio system will my opinion change if I'm presented with a set of impeccable measurements?

Extremes

Must we choose between these extremes or is it possible to combine the best elements of both objective and subjective evaluation and arrive at the most satisfactory conclusion?

I believe that it is not only possible but absolutely necessary in order to reach musical listening nirvana. I think we need to select those audio components and systems that provide emotional satisfaction from among those that are demonstrably accurate according to accepted standards. Accepted standards include, but are not limited to, reasonably accurate measured performance.

Yes, Virginia, Facts Exist

Facts are simplistic, but they can be very useful. They don't necessarily demonstrate the value of something but they sure can help to expose a fraud. This **Journal** will explain speaker measurements and show their value. This knowledge won't provide a substitute for personally auditioning components but it will narrow the field.

Objective measurements can't tell you everything about how a component or system sounds but they absolutely can tell you when the search for accurate reproduction is hopeless.

Why Are Certain Facts Important

In this issue I'll explain why certain performance characteristics are important for the accurate reproduction of recorded music. These articles will offer my opinions and illustrate the logic that led me to these conclusions. Then I'll explain how to tell which transducers provide these characteristics and which ones can't possibly present an accurate reproduction of the recorded information.

How Do You Tell Which Products Work?

After each of the articles that describes desirable performance attributes there will be an article that explains how these attributes can be measured and how you can interpret the measurements. Yes, objective facts exist and they can be of value to you.

Also In This Issue

Shane Buettner reviews the Thiel CS2.4, an outstanding, high-value speaker system. And we'll present my interviews with Jim Thiel and Richard Vandersteen. APJ

Why Accurate Speakers Must Have Flat Frequency Response

by Richard Hardesty

All audio components, including loudspeakers, must have reasonably flat frequency response in order to accurately reproduce the timbre of musical instruments and voices. This fact is clearly evident and here's why.

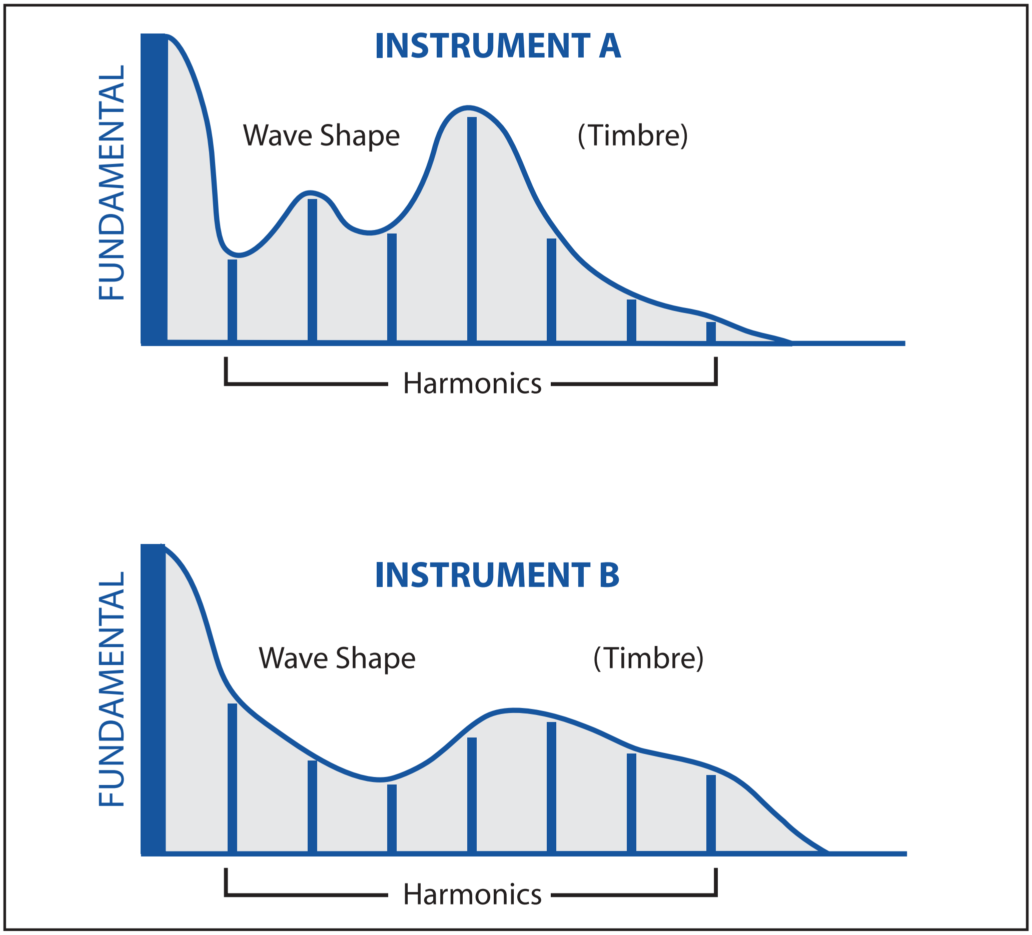
Every sound produced by a musical instrument or voice is made up of a fundamental frequency and a series of harmonic overtones. The fundamental frequency defines the note being played. The harmonic structure allows the listener to identify the specific instrument playing or voice singing the note.

The complete sound of each note the listener hears is a combination of fundamental and harmonics. The fundamental establishes pitch and the harmonic structure defines the unique characteristics of each instrument or voice based on tone rather than pitch or volume. These unique tonal characteristics are called timbre.

> ***"…instruments sound different because each has a unique harmonic structure (timbre)…"***

A piano and a violin can play the same note with the same fundamental frequency. The pitch and volume of the notes from the piano and violin may be identical. The instruments sound different because each has a unique harmonic structure (timbre). The waveforms from these instruments have different shapes when displayed on an oscilloscope and each delivers a unique sound to the ear even when playing the same note at the same volume. The fundamental frequency of that note is identical but the harmonic structure is different. The tonal characteristics (timbre) are different and easily identified by the ear.

The unique timber of each instrument or voice is established by the quantity and ratio of harmonic overtones which determine the shape of the wave. The exact relationship of these overtones must be preserved in order to reproduce timber accurately. This requires that audio components exhibit flat frequency response and maintain proper timing relationships. Flat frequency response is required in order to preserve the amplitude



sound. A speaker system that delivers midrange harmonics in reverse phase can't possibly recreate musical timbre accurately yet that's exactly what most speakers do.

The information that follows will explain how to tell which speakers are correct in the amplitude and time domains. While the articles in this **Journal** will be specifically about speakers, the facts apply to all audio components. Speakers serve as good examples because they tend to be the least accurate of components and exhibit the grossest deviations from accurate amplitude response and phase. APJ

relationships between the fundamental and each harmonic overtone. Proper timing must be maintained in order to preserve the phase relationships between the fundamental and harmonic overtones.

If the speaker system or other component deviates from flat response, signals at some frequencies will sound louder or softer than they should. These response deviations may alter the level of various harmonics and change the timbre of the sound. Some overtones may be emphasized and/or some may be diminished in level. A speaker system with response deviations of just a few decibels will alter the timbre of musical instruments and voices. A speaker system with response that deviates beyond a window of ±3dB can't possibly recreate the timbre of instruments and voices accurately.

If a speaker system or other component alters the phase of any group of frequencies, the wave shape—and the sound—will be altered. Slight changes in phase will slightly alter the timbre of the sound and phase reversals will substantially alter the

How Can You Tell if a Speaker System has Flat Frequency Response?

by Richard Hardesty

You can test speakers yourself with readily available computerized instruments like Clio (www.mclink.it/com/audiomatica/home.htm) and MLSSA (www.mlssa.com/) or you can refer to the measurements printed in trusted publications.

John Atkinson of Stereophile magazine, for instance, has measured hundreds of speakers. He uses methods generally accepted by the industry and achieves useful results. Stereophile speaker reviews are available free on the web at www.stereophile.com and they can show you that most of the speakers that Stereophile reviewers recommend are incapable of accurate performance.

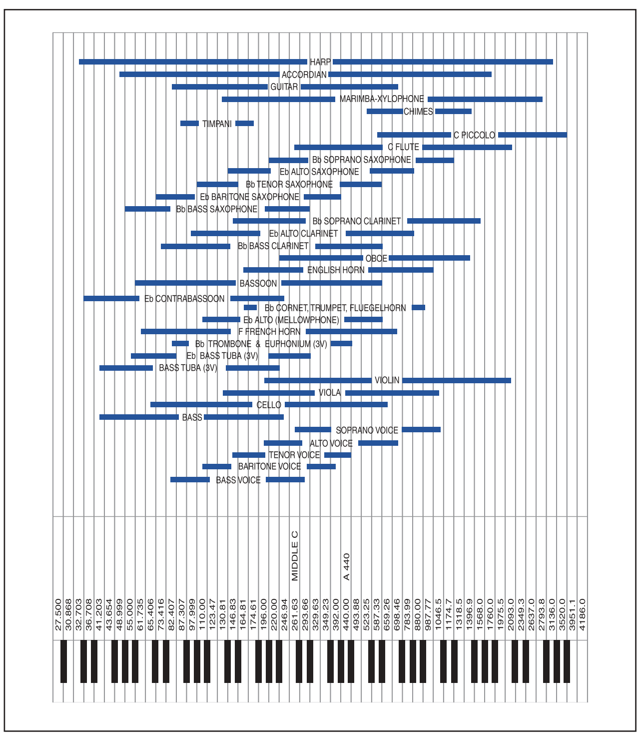
The frequency response graph will tell you what you need to know about the ability of the speaker under test to deliver an accurate replica of the recorded information in terms of amplitude. Flat frequency response is the cornerstone of good speaker design and must be considered before any other parameter but it's not the only important factor, as we'll see.

A frequency response graph displays frequencies on the horizontal scale across the bottom, sometimes called the "X" scale, and amplitude on the vertical scale at the left side of the graph, sometimes called the "Y" scale. You can use this graph to determine if the speaker under test deviates from flat response, and to pinpoint the amplitude and frequency of that deviation. Don't be intimidated, it's very easy to do. All you need to know is what the gradations on the graph mean and how to interpret that information. I'll try to explain that in the simplest terms possible.

Frequency

The frequency scale across the bottom is usually logarithmic (abbreviated log) rather than linear, which means that the frequency goes up in powers of ten—ten, one hundred, one thousand, ten thousand.

On a logarithmic scale more space is devoted to octaves at the bottom of the scale than to those at the top, which correlates well with the actual frequency ranges of musical instruments



(see **Journal #12**). Frequency is shown in cycles per second, now called Hertz (abbreviated Hz). Hertz and cycles per second are two names for the same thing—100Hz is 100 cycles per second. One thousand Hertz is often called a kilohertz (1kHz).

The frequency range of an engineering instrument may be from 10Hz to 30kHz. The range displayed by instruments often used by reviewers may be limited to 20kHz. The 20kHz upper limit covers the range of human hearing but can't show ultrasonic tweeter resonances, which can have an audible effect on the frequencies we can hear. The "oil can" resonance of an alloy dome tweeter will typically occur at frequencies of 25kHz to 27kHz, which won't be displayed by a test instrument with a 20kHz bandwidth but may affect the sound of the speaker under test.

Amplitude

The vertical scale at the left side of the graph shows amplitude in decibels (abbreviated dB). The decibel scale is logarithmic, not linear, and this is important to remember. This scale may show numbers in a range of perhaps 60dB-95dB, or it may have a point marked zero with ranges graduated in decibels above and below this point so you can determine how far the speaker under test deviates from flat response.

This scale indicates the volume of the output from the speaker under test. We're not concerned with the actual volume of the output. What we want to know is the relative volume across the frequency scale so we can see which frequencies are emphasized and which are diminished and by how much.

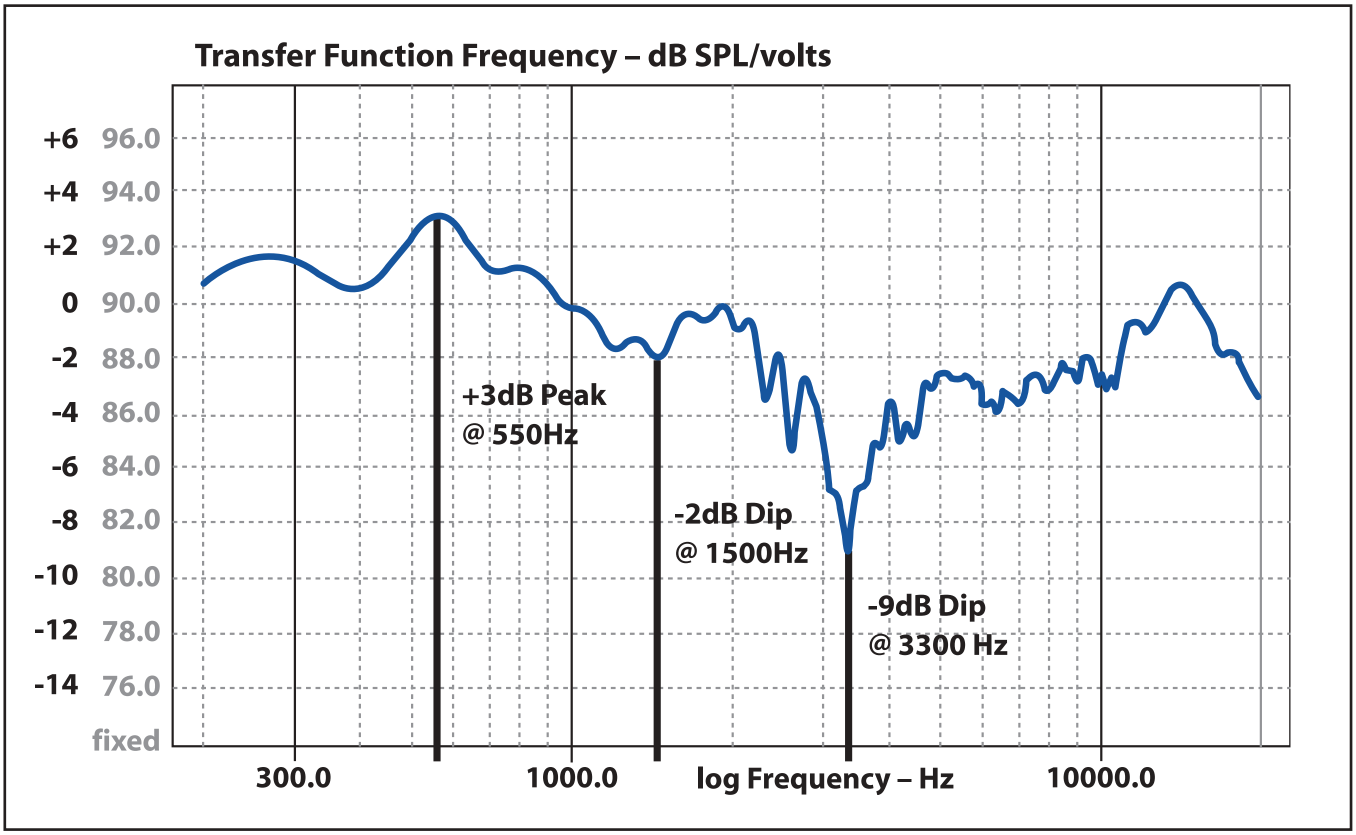
The Stimulus

The test signal is called a stimulus and speakers are typically stimulated to produce an output of 80-85dB SPL at 1kHz but that doesn't matter much. As noted above, what we're interested in is how much louder or softer the speaker plays at various frequencies with an input signal at a constant level.

Most speaker measurements use quasi-anechoic techniques to eliminate the effects of the room in which the speaker is measured. The test signal for a quasi-anechoic measurement is called a Maximum Length Sequence (MLS), sometimes referred to as pseudo-random noise, which is converted into an impulse by a computerized test instrument. The impulse is then converted by the computer into frequency and phase information utilizing the Fast Fourier Transform (FFT), which is a mathematical method of converting between time domain and frequency domain information.

You don't have to understand how this works in order to realize the limitations and benefit from the results. (If you do want to understand read *Testing Loudspeakers* by Joseph D'Appolito from Audio Amateur Press.)

been spliced together with the frequencies above about 500Hz measured at a distance of three or four feet using quasi-anechoic techniques, and the frequencies below 500Hz created by combining a close-microphone measurement of the woofer(s) and a close-microphone measurement of the vent(s) using mathematical techniques to add the outputs at appropriate frequencies and calculate the difference at frequencies where the driver(s) and vent(s) would be out-of-phase.



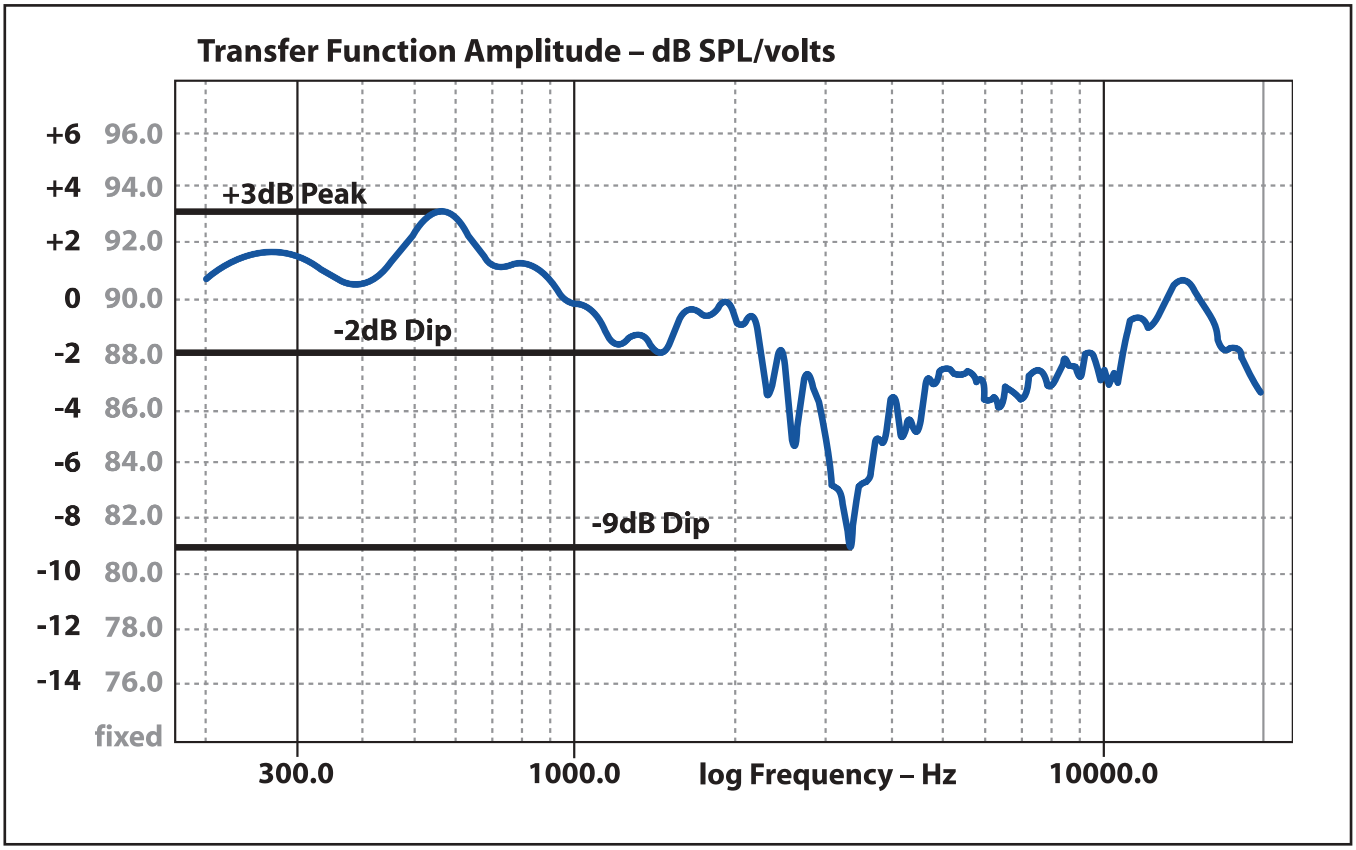
Quasi-anechoic measurements minimize the effects of ambient noise and reflections from the room boundaries and allow speakers to be tested in regular rooms rather than anechoic chambers. The measurement window starts when the first sound arrives at the speaker and ends with the arrival of the first sound reflected from the room boundaries. The room boundary nearest to the test microphone will usually be the floor, which limits the accuracy of low frequency measurements.

If the speaker under test is standing on the floor the accurate lower limit of the measurement will be about 500Hz. If the speaker is elevated that limit may be extended to 300Hz or slightly below. The measurements you see in magazines have

The Graph

The "X" and "Y" scales together show how loud the speaker plays at each frequency in response to a test signal with a constant volume or one that can be correlated to a constant volume. A perfect speaker would produce a straight line across the frequency range at 80-85dB on a scale marked in dB, or at the zero line on a scale that shows zero with ranges above and below that point graduated in decibels. An imperfect speaker will produce a horizontal line with small deviations above and below the zero line (or a horizontal line drawn through the amplitude point at 1kHz).

A deviation above the zero line means that the speaker under



test will emphasize signals at that frequency. A deviation below the line means that signals at that frequency will be diminished by the speaker under test.

You can determine the amplitude of the deviation by following a horizontal line from the tip of the deviation to the scale at the left side of the graph. Be sure to note whether the scale is graduated in 5dB increments (typical) or other increments. You can determine the frequency of the deviation by following a vertical line down to the scale at the bottom of the graph. (See the illustration.)

Deviations are usually referred to by the difference in amplitude between the level at 1kHz (1,000Hz) and the level at the deviation. If the specification says frequency response is 40Hz-18kHz ±3dB, that means that when tested the response of the speaker never deviated more than 3dB from flat response between the speaker's bass limit at 40Hz and the highest treble it could reproduce at 18kHz. In other words no output peak was more than 3dB louder than the signal at 1kHz and no output dip was more than 3dB softer than the output at 1kHz. This 6dB window of error over this bandwidth is the minimum performance that I would consider acceptable for high fidelity music reproduction.

The Real World

Speakers that utilize D'Appolito arrays (two midrange drivers, one above and one below the tweeter) will measure better than they sound because the microphone will be exactly centered between the midranges and the response differences of the two drivers will produce a smoothing effect on the frequency response. In the real world a listener will never be exactly centered and the resulting time smear will simply blur definition.

Speakers with dipole radiation patterns will sound better than they measure because the horrendous response deviations in the direct signal will be smoothed by the rear-wave energy reflected off the wall behind the speakers. In the real world this time smear will prevent any real definition but will provide an artificial ambience effect that some find pleasing.

Speakers with bipolar radiation will create even more time smear than those with dipole radiation in most rooms because they radiate substantial energy to the sides increasing the amount of reflected sound that arrives at the listener. Some people like this sound but it's not related to the recorded signal.

There's More to Accuracy

A speaker that can't produce a fairly accurate frequency response graph can't reproduce an accurate replica of the recorded information but there's much more. Phase and energy storage are separate issues that will be considered in the articles that follow. APJ

Why Must an Accurate Speaker System be Time- and Phase-Correct?

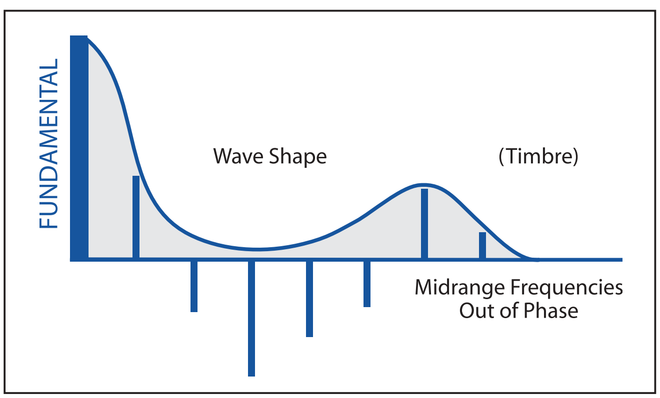
by Richard Hardesty

As described in the articles about frequency response, correct timbre cannot be reproduced by a speaker system that alters the amplitude of any harmonic overtone because that will surely alter the shape of the resulting waveform. The same is true of phase. Reversing the phase of portions of the frequency spectrum changes the shape of the waveform. No audio component—except the loudspeaker—does this.

Stereo imaging is of great importance to some (like me) and couldn't matter less to others. This effect has a more subtle influence on musical realism than the accurate reproduction of timbre but is significantly impacted by the time domain performance of the speakers.

Timbre

The harmonics of real instruments or voices add to or subtract from the fundamental frequency which creates pitch. This results in unique tone called timbre that allows us to identify the instrument or voice. A piano doesn't sound like a violin and Sheryl Crow doesn't sound like Ray Charles even when they each play or sing the same note. The pitch is identical but the timbre—and the waveform—is different.



Arguing about whether an alteration to the shape of the resulting waveform is audible is like arguing about whether a frog is waterproof. Observation provides irrefutable evidence. You can observe frogs in a pond and you can observe waveforms on an oscilloscope. A frog emerges from water completely unscathed and musical instruments produce different waveforms on an oscilloscope. Changing the phase of any wave component alters the shape of that waveform. Period.

Of course we can distinguish between waveform shapes with our ears. How else could we hear timbre and distinguish between various voices and instruments? Helmholtz demonstrated this fact 150 years ago and modern day musicians are well aware of it even if they can't necessarily provide an explanation. Piano tone can be altered by changing the point where the hammer strikes the string because this alters the relationship between the harmonics that string produces. If the piano is in tune the fundamental frequency will not change, regardless of where the hammer strikes the string. If the piano is not in tune it can be adjusted with an electronic tuning device that displays the fundamental frequency (pitch) of each string. The same tuning device can be used with other instruments even though they have unique timbre.

Guitar players can alter the tone they produce by changing the place where the string is plucked or altering the way in which

the string is set in motion (nail, pick or flesh). If the guitar is in tune the pitch stays the same. The switch at the bottom of a Fender Stratocaster changes the electrical phase relationship of the three pickups positioned beneath the strings. This 5-position switch alters the tone of the instrument, not the pitch, which is established by the fundamental frequency of the vibrating string.

Phase does matter and those who suggest that it doesn't probably make speakers which are not time coherent.

Imaging

The effect of phase on stereo imaging is more subtle but can be observed with experience. Time- and phase-accurate speakers produce a stage that is rectangular (as viewed from above) with depth that is apparent well to the sides of that stage and not only at the center. Stage width can extend well beyond the loudspeakers.

Typical speakers produce a stage that is triangular (as viewed from above) with an illusion of depth only at the center and a reduced sense of depth toward the speakers (sides of the stage). Time incoherent speakers will usually define the outer edges of the soundstage. APJ

How Can You Tell if a Speaker System is Time- and Phase-Accurate? *by Richard Hardesty*

Speaker manufacturers and reviewers have presented loads of meaningless misinformation about the importance of phase and you've probably seen dealers or manufacturers meticulously adjusting the position of drivers or speaker tilt-back, as if this has an effect on phase coherence. It doesn't.

Phase shift is primarily the result of crossover filters which divide the spectrum into ranges of frequencies which are directed to appropriate drive elements. Phase shift varies with frequency and can't be corrected by changing the physical position of the drivers.

Driver position can affect the time relationship between drivers because each has a slightly different rise-time and driver position may have a minor effect on cancellation in the overlap region where two drivers are reproducing part of the same signal which has been phase-shifted by the crossover filters.

Speakers with steep filter slopes can't be made time- and phase-correct. Speakers with first-order acoustic slopes will also require that drivers be carefully positioned to compensate for differing rise-times. Read this paragraph again because it's very important.

A time- and phase-accurate speaker will have gentle filter slopes and physically aligned drive elements. One without the other won't do and all talk to the contrary is simply rhetorical. Time and phase do matter and there is a sure way to tell when these timing relationships are correct.

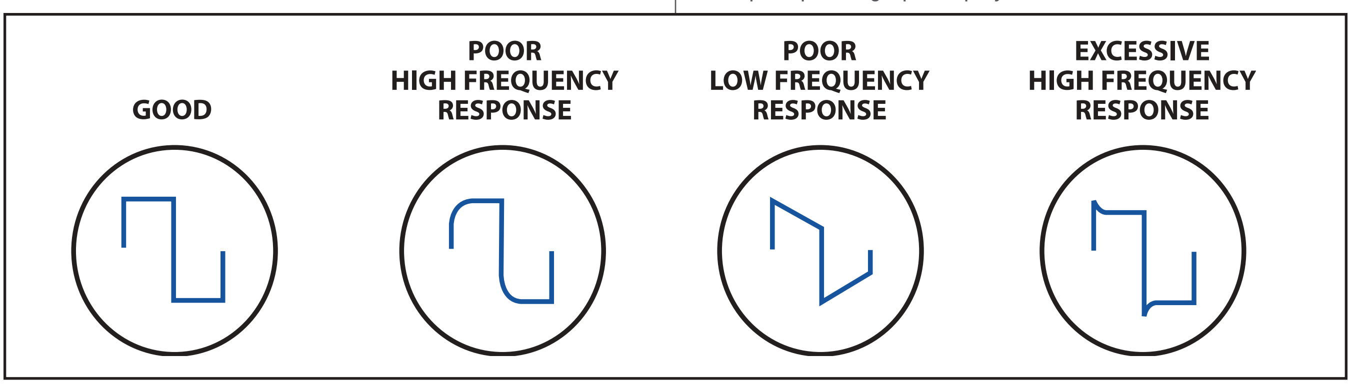
When the Green Flag Drops the Bullshit Stops

The step response graph tells whether a speaker system is time- and phase-accurate, clearly and without ambiguity. If some drivers pull while others push, the speaker is not phase coherent and this will be revealed by the speaker's step response.

A speaker that is not phase coherent cannot accurately reproduce timbre or image correctly. The shape of the step response graph tells all. Here's why.

The Step Response Graph

A step response graph displays time on the horizontal scale



across the bottom, sometimes called the “X” scale, and amplitude on the vertical scale at the left side of the graph, sometimes called the “Y” scale. You can use this graph to determine if the speaker under test is time coherent or if some drivers push while other pull.

The factor important to us is the shape of the output signal, not the increments which may be significant to designers but not to listeners. The step stimulus is like the top half of a square wave or an impulse with extended duration. I’ll try to explain that in the simplest terms possible.

The Square Wave

I’ve heard it said that we don’t listen to square waves but the lowly square wave is the test signal that most resembles music. A square wave, like a musical wave form, is composed of many sign waves—in fact an analog square wave generator creates a square wave by generating a sine wave at the fundamental frequency and adding sine waves for all the odd harmonics. The resulting square wave is a combination of many sine waves with precise amplitude and phase relationships, just like music. If everything is right the result looks square. Deformation indicates deviation—showing that some part of the signal has been altered.

You can learn almost everything about amplifier performance by observing the square wave response. Most speakers can’t even come close to reproducing a square wave because some drivers push while others pull and there can be no output when a drive element is stationary. To gauge the time domain performance of loudspeakers we use the step response instead.

The Test Signal for the Step Response

The step response is actually derived from an impulse correlated from an MLS stimulus but for understanding think of it like an impulse with extended duration or the top half of a square wave. The signal starts at zero, rises quickly to a level that is maintained for a time at a positive DC potential and then returns to zero. The signal never goes negative, which is important to remember. This signal tells you which drivers move in which direction and when, but the speaker can’t produce a step that exactly matches the input signal because there is no sound when the drivers reach maximum excursion in response to a constant current (DC). Sound occurs as a drive element moves and stops when the drive element stops, even if it stops at the end of a long excursion. Why? Because a stationary driver diaphragm can’t move air and create sound.

If you apply direct current (DC) to a loudspeaker the woofer will be displaced in direct proportion to the amplitude of the current—and stay there until the current is removed. The capacitors that act as high-pass filters for the other drivers will block DC allowing only the upper harmonics to pass. The stimulus exercises the entire speaker just like music, but only momentarily. With computerized test instruments that’s all we need to determine which drivers move to create positive displacement (outward) and which ones are out-of-phase (move inward). These facts are important to know if we want the output waveform to mimic the input signal, which is absolutely necessary for accurate reproduction of the recording.

Step Response

The step response graph shows only the output of the speaker under test. Think of the stimulus like the top half of a square wave. The output from a time- and phase-accurate speaker should look like a triangle above the reference line with a sharp rise and a slow decay. The beginning rise will slope slightly because the step stimulus rises almost instantly (straight up) but the speaker has limited bandwidth and takes some time to rise. (Bandwidth and rise-time are corollaries.)

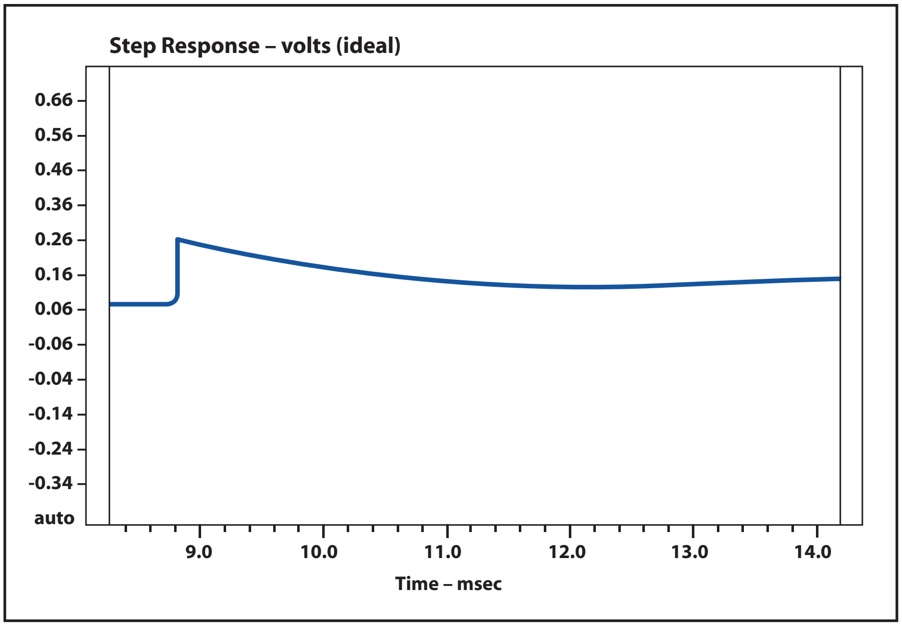
The speaker makes sound as the drivers respond to the stimulus and then output ceases so the signal on the graph decays back to zero, and maybe a little beyond due to inertia (rebound), over a period of a few milliseconds.

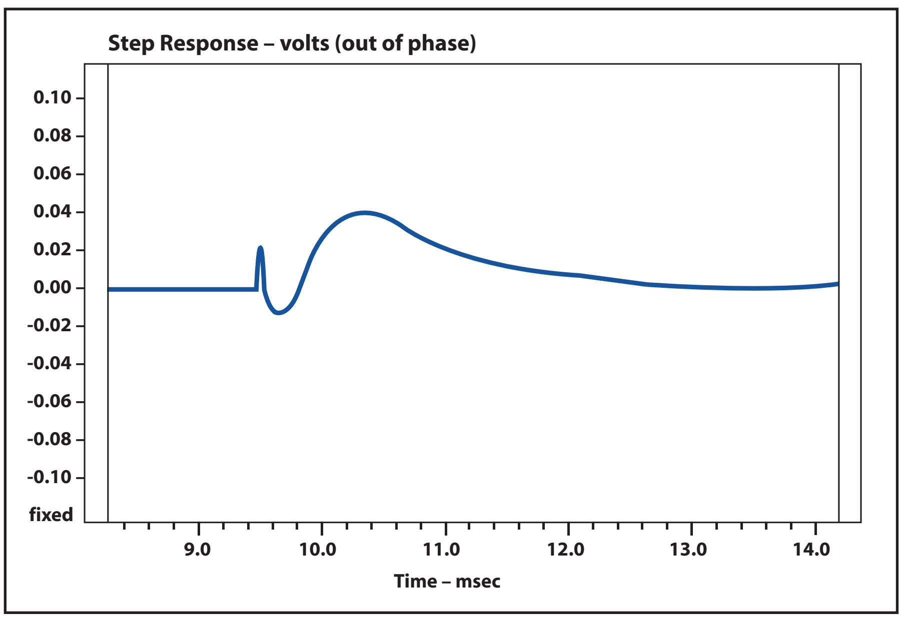
The test signal never goes negative so any significant output that extends in a negative direction (below zero) is out-of-phase. You can’t tell for sure which part of the frequency spectrum is out-of-phase but the output from the tweeter usually arrives first followed by the output from the midrange (if there is one) and then the woofer.

A speaker with a third-order crossover will typically have the midrange driver wired out-of-phase. A speaker with a fourth-order crossover will typically have the drivers wired in phase but their output will be smeared over time and a portion may still go negative due to crossover phase shift.

Examples

The following examples show an ideal step response and the step response from a typical speaker. A step response graph that is not a positive (upward) triangle indicates a speaker that is not time- and phase-accurate. Trying to extract additional data from the graph is not within the scope of this article or the experience of most readers.





Don't be fooled by the appearance of a speaker or rhetoric from the manufacturer. The step response graph shows which speakers are time- and phase-correct and which ones aren't. Most of them aren't. APJ

Why Must Energy Storage be Minimized?

by Richard Hardesty

Speakers contain electrical and mechanical components that tend to store and release energy after a short delay in time. This tendency must be minimized if the speaker is to reproduce an accurate replica of the recorded information and nothing else. What we want to hear is the recorded information. We don't want to hear sounds created by the speakers or sounds that originated from the recording but have been delayed and reradiated by the speakers.

Musical instruments are designed to deliver rich and pleasing resonances. Accurate loudspeakers are designed to reproduce recorded sounds rather than create sounds of their own. A recording should capture resonances produced by real musical instruments and speakers should accurately reproduce the recording and nothing else.

The audible output from stored energy can be attenuated by damping and what remains will be released after a delay. The amount of attenuation will be determined by the damping qualities of the resonating material. The delay will be determined by the amplitude and frequency of the stored energy. This reradiated (stored) energy will produce inappropriate sounds that may blur or obfuscate the actual signal.

Perfect speakers will deliver the recorded information at the right time and then stop producing sound. Imperfect speakers will deliver the recorded information and store some energy that will be released later when there should be silence or when other recorded information should be produced. Nothing is perfect, of course, but well-designed speakers will reduce stored energy to an absolute minimum.

Sources of Stored Energy

Resonant energy can be stored in enclosure panels and driver diaphragms. Electrical energy can be stored in crossover components. Reflected energy can be radiated by enclosure baffles, driver frames and internal enclosure structures. Some of these sources will subtly degrade the signal and some will have a profound negative impact on sound quality.

> ***"A good speaker design will address all these sources of stored energy and reduce each one to inaudibility."***

A good speaker design will address all these sources of stored energy and reduce each one to inaudibility. A poorly designed speaker will deliver a signal that combines part of the recorded information with the stored energy from the speaker itself. The frequency response graph may appear to be flat with dips in the signal reproduction filled by resonant peaks created by speaker components and vice versa. That's why no single measurement can be trusted and even a group of measurements must be viewed with skepticism. The measurements can be useful to narrow the field of contenders but must not be used as a substitute for careful listening evaluations.

Mechanical Resonances

Enclosure panels can resonate and this tendency can be minimized by making the panels thicker; constructing them from dense, well-damped material; adding cross braces to larger panels to raise the frequency and reduce the amplitude of the resonance; and/or constructing the panels from exotic materials, or laminates that contain exotic materials, with high internal damping characteristics.

Driver diaphragms can be constructed from materials with high internal damping and/or drive elements can be used only to reproduce frequencies that are substantially below (or above) the diaphragm resonant frequency. This is particularly difficult with first-order speaker systems where each driver must cover a wider frequency range than would be necessary in a speaker with steeper crossover slopes. (See **Journal #12**).

Electrical Resonances

Capacitors and inductors are used as high-pass and low-pass filters in speaker crossover networks. Capacitors resist a change in voltage and inductors resist a change in current. Both elements cause a phase shift between voltage and current.

In an inductive circuit voltage leads current by about 90 degrees. In a capacitive circuit current leads voltage by about 90 degrees. Certain values can result in resonant electrical circuits. Generally, a crossover network with fewer components is better but many crossover components may be used to compensate for driver anomalies.

The crossover networks in well-designed speaker systems have been carefully engineered to divide the frequency spectrum and direct the divided bands to the appropriate drivers; compensate for driver anomalies in frequency and phase; and provide a constant and acceptable load to the amplifier. This is a very tall order.

If you think you can build a speaker system in your garage that is competitive with today's finest designs, think again.

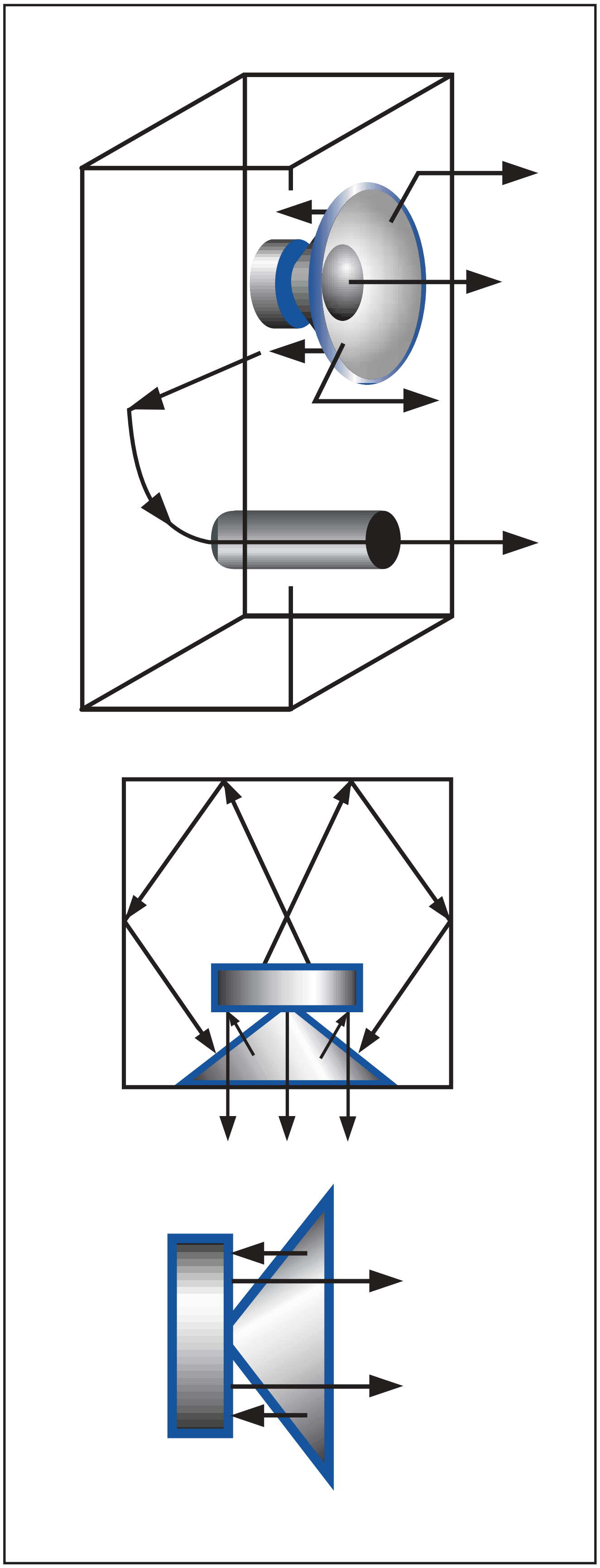
Reflected Energy

The most prominent source of reflected energy is the baffle that surrounds the speaker drive elements but there are others. Energy radiated from the back of a drive element can be reflected by cabinet structures and reenter the room through a woofer port. This rearward radiation can be reflected by the magnet structure and frame of the drive element and reenter the room through the diaphragm that produced it, delayed by the additional path length from the diaphragm to the structure and back.

Regardless of the source, reflected energy travels farther than direct energy and smears the signal over time, blurring definition and degrading imaging.

Stored Energy is Bad

Some speakers are designed to create artificial sounds by storing energy in shimmering Mylar™ panels or purposely directing energy toward the walls where it will be reflected back to the listener. Some people enjoy this artificial "ambience" but it's not related to the recording and can't be called high fidelity reproduction by any stretch of the imagination. APJ



How Can You Tell if a Speaker Stores Energy?

by Richard Hardesty

I use the phrase "energy storage" to describe a variety of flaws which result in time smear. Resonances and reflections fall into this category. Stored energy is a major source of coloration and a primary reason why speakers that measure well in other areas sound different.

It's easy to hear these flaws but the exact source of the problem is often difficult to pinpoint and correct. That's why we still get a broad range of sound from speakers which appear similar in design and have reasonably accurate frequency response.

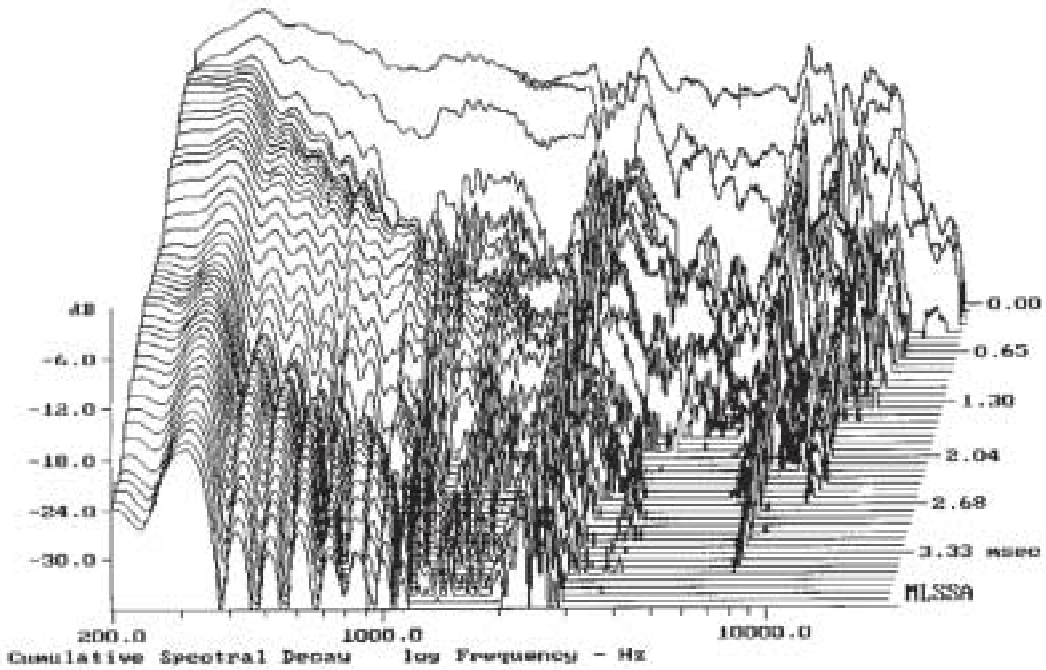
An amateur builder can order excellent drivers from Denmark and use a computer program to design a crossover network and still end up with a loudspeaker with substantial colorations. There are some very vocal "geniuses" on the internet who would have you believe that speaker design is a simple matter that has been reduced to mathematics. They're wrong. A good engineer will consider all aspects of his design and energy storage encompasses several important factors. You'll want to consider only those speakers which were designed by good engineers.

Measuring Stored Energy

Energy storage is not as simple to measure as other parameters but you can get some idea from spectral decay plots and accelerometer measurements of cabinet resonances. The spectral decay plot shows which frequencies linger and for how long but the source of these lingering frequencies is not easy to identify. An accelerometer attached to a cabinet panel can show the resonant energy stored in that panel, identify the frequency of that resonance and show if it's well controlled or potentially audible.

> ***"The spectral decay plot shows which frequencies linger and for how long…"***

An accelerometer may spotlight a panel resonance but can only suggest how deleterious this resonance may be to the sound from the speaker so you'll have to use common sense



This is an example of the poor cumulative spectral decay plot produced by a planar-magnetic speaker. The graph starts with the line at the top. Each line below that shows the spectrum of response at a later time. This graph clearly displays critical midrange energy that persists for several milliseconds after the signal ceases. Some listeners will mistakenly interpret this as "ambience" but the source of this energy is the speaker, not the recording.

during final evaluation by listening. Designers have to try to solve problems. Consumers can simply avoid them.

Problems at the front of the speaker will tend to be more audible than problems at the sides and rear. Stored energy at lower frequencies will tend to blur details more and stored energy at higher frequencies will tend to be more directly audible. Here's an explanation of the graphs in the simplest terms possible.

The Spectral Decay Plot

The spectral decay plot is frequently called a "waterfall" because of its appearance. It displays three parameters—time, frequency and amplitude—in a two-dimensional representation of a three-dimensional graph. This graph is slightly more complicated than the ones that show just two variables but not as difficult to interpret as it first seems.

Frequency is displayed along the bottom usually in log form. Time is displayed at the right side usually in milliseconds. Amplitude is displayed on the left side of the graph usually in dB. An amplitude level like -24dB means that the signal at this point is 24 decibels softer than the original signal at the top (uppermost trace). Instruments frequently "autorange" (self adjust levels) so pay attention to the amplitude levels displayed. They won't necessarily be the same for all graphs.

The graph consists of a series of wiggly horizontal lines. The one at the top is the first one, which shows the response to the test signal, and each line below that is the response at a slightly later time, as indicated on the time scale at the right.

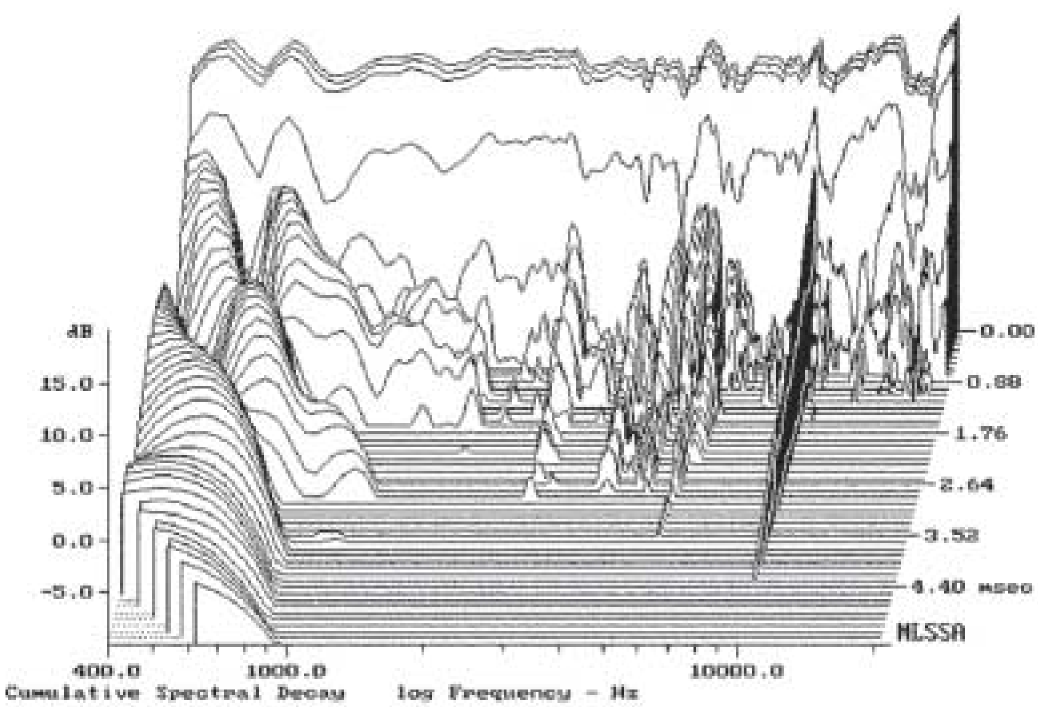
The wiggly horizontal line at the rear (top) shows what is essentially the frequency response of the speaker under test and each successive line toward the foreground (bottom) is the spectrum of response after a period of time has elapsed. These traces show which frequencies emanate after the signal has stopped and how long they persist. A ridge indicates substantial stored energy, probably from a resonance.

> ***"A real speaker will deviate from flat response and keep singing after the song has ended…"***

A perfect speaker would produce a nearly horizontal line at the top and nothing else because it would store and release no energy after the signal ceases. A real speaker will deviate from flat response and keep singing after the song has ended—continuing to produce sound for a period of time after the signal has stopped. A well-designed real speaker will produce little sound after the signal stops and that sound will be well down in amplitude so it won't intrude on the signal from the recording. So how do you gauge the performance of the speaker under test? Examine the spectral decay plot.

Interpreting the Spectral Decay Plot

The top of the graph is offset slightly to the right to show the three variables in a simulation of three dimensions. The test instrument will allow the tester to place a curser at the point of a deviation and read the frequency directly but you'll have to follow a vertical line to the bottom of the graph (moving slightly to the left) to determine the frequency that is being sustained.



This is an example of a pretty good cumulative spectral decay plot. With the exception of a fairly prominent resonance at about 10kHz, residual energy is greatly reduced in level and virtually gone after an interval of less than 2.5ms.

You can determine the duration of the sustained frequency by following a horizontal line to the time scale on the right side of the graph. The graduations here show the amount of time which has elapsed since the initial trace. Two milliseconds is a long time in speaker terms and a substantial output after that length of time indicates an audible coloration, which would be unacceptable to me.

You can determine the amplitude of the sustained frequency by following a horizontal line to the amplitude scale at the left. The graduations here show how much the amplitude of a deviation has diminished since the initial trace. Sounds that linger after the signal stops should be well down in level when compared to the original trace, perhaps reduced in level by 24 to 30dB after a millisecond or so has elapsed.

In a typical quasi-anechoic measurement setup, frequencies below 400Hz or so will not be accurate. If the measurements were taken at a greater distance in a large anechoic chamber accurate bandwidth may be extended down to 200Hz or so. Anything below this frequency should be ignored.

Accelerometer

A cumulative spectral decay plot (waterfall) can be made using an accelerometer instead of a microphone as a sensing device. Inexpensive accelerometers are readily available and can be attached to cabinet panels. The output from the accelerometer is fed to the test instrument instead of the output from a microphone, as in most tests. The resulting measurement will display any prominent panel resonances, identify the frequency and suggest the extent of the problem.

Determining how this will affect the sound from the speaker under test is a subjective matter best left to experienced testers. You can see from the graph when there may be a problem and listen to determine if that problem is sufficiently audible to be problematic to you. APJ

The Importance of Bass

by Richard Hardesty

You may have heard that clear midrange is all that's necessary for a satisfying experience from recorded music. If you believe this I can make you a great deal on a large bridge in San Francisco. Midrange isn't an entity unto itself. It exists partly as a result of bass and treble. The idea that bass plays a small roll in music is simply not true. Bass is more than a minor attribute of music—it's a vitally important component.

There is a small kernel of truth in the big bass lie: you probably are better off without the poorly defined and badly timed bass produced by most speakers. But a true aficionado will never be satisfied by reproduced music with poor bass. Bass is the foundation of all music and a major component in both of the factors that separate music from noise: the rhythm (beat) and the melody (tune).

A percussion instrument like a drum often sets the rhythm and establishes the pace of a piece of music. Bass instruments add body to this rhythm while playing a tune in harmony with the instruments that create the melody. The upper harmonics produced by bass instruments provide a substantial contribution to the melodic content of the music, which is typically perceived as "just" midrange.

Real Bass

We're not talking about "car stereo" bass, which is really emphasized mid-bass. We're talking about real low frequency fundamentals below about 80Hz. These bass frequencies are an important part of a live musical performance and must be present for a reproduced performance to sound like a reasonable facsimile of the original. You can hear music without bass over the telephone but it's only a vague suggestion of the real thing.

To fully convey the musical message, bass must be present and it must be tightly controlled and presented with impeccable timing. (Refer to **Journal #12** for the actual frequency ranges of real musical instruments.) I can't be satisfied by an audio system that lacks outstanding bass performance and once you've heard a system that presents bass correctly you probably won't be satisfied with less either.

You can subjectively evaluate bass performance by observing your rhythmic involvement with the music. Good bass makes it easier to follow the beat and it makes the band seem to be playing together in tighter unison. Percussive elements define and punctuate the experience.

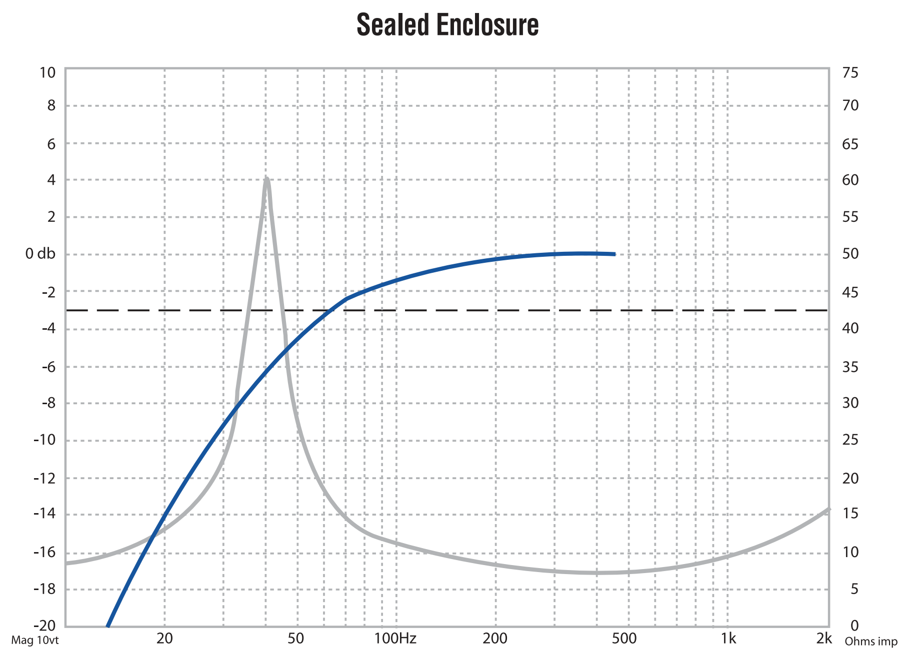
You can subjectively evaluate the contribution of bass frequencies to the overall presentation. With good bass, music sounds fuller and more balanced. Bass harmonies add complexity and interest to the musical score. Deep bass notes can deliver a thrilling comment to an otherwise subtle tune.

It's more difficult to objectively evaluate bass performance but it is possible to gain some vital knowledge by examining measurements. Let's learn how. APJ

Evaluating Bass Performance

by Richard Hardesty

It is possible to accurately measure low frequency performance outdoors (even very large anechoic chambers are too small) but this is seldom practical for magazine reviewers. What you'll see in most reviews is a measurement made by placing the microphone very close to the radiating diaphragm or vent opening and splicing this measurement, or the complex sum of these measurements if there is a vent, to the quasi-anechoic graph that shows frequency response from 400Hz or 500Hz on up.



The blue line is frequency and the gray line is impedance.

This low frequency graph gives a pretty good idea of the speaker's response with no indication of the room's effects. I've found that these measurements correlate well with what I hear when auditioning speakers tested this way. You can get additional information by examining the impedance plot.

Impedance Plot

The impedance plot tells us a great deal about the design of the woofer section of the speaker and reveals clues about transient response. Of course it shows a number of other interesting things that are beyond the scope of this article, but we'll use it as part of our bass performance evaluation.

The number of resonant peaks in the impedance plot and the frequency of these peaks indicate the type of bass loading, as does the rate of low frequency roll-off. The speakers we are likely to encounter will have sealed enclosures or vented enclosures.

The vented designs will probably have tuned ports, passive radiators or transmission line loading.

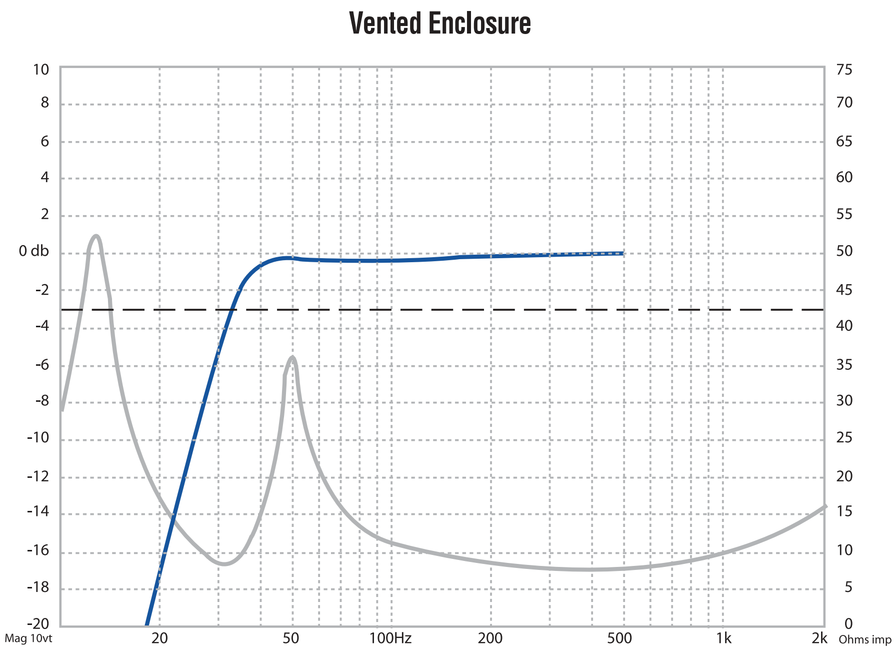
Sealed Enclosures

A sealed enclosure will typically have a single resonant peak, which will coincide with the –6dB point or low frequency limit of the speaker under test. Output from the speaker will roll off at about 12dB/octave below this point. A well-designed transmission line will behave more like a sealed enclosure than a vented design. It will probably have a single resonant peak with reduced amplitude as compared to a sealed enclosure. The resonant peak will probably occur at a slightly lower frequency as compared to a sealed enclosure.

Vented Enclosures

A vented enclosure will have two resonant peaks and the bottom of the trough between them will define the resonant frequency of the air mass in the vent. The resonant frequency of the vent will coincide with the minimum excursion point of the bass driver demonstrating that what you hear at this frequency is the vent resonance not the output from the bass driver.

> ***"…an impedance plot with two peaks is the least desirable indicator of bass performance."***



The blue line is frequency and the gray line is impedance.

Output from the speaker will roll off at about 24dB/octave below this point.

Passive Radiators

A design with a passive radiator will behave very much like a vented design. A band-pass design will behave like two filters—one at the bottom, which could perform like a sealed enclosure if one side of the band-pass woofer is sealed, and one at the top, which will perform like a vented enclosure in reverse, providing a fourth-order low-pass filter.

Impedance Peaks

Two impedance peaks are the least desirable indicator of bass performance. Two peaks are produced by a vented design with the steepest roll-off and the poorest transient response. One peak is better, indicating a sealed design with more extended bass response and improved transient performance. No peak is best, indicating a design with little or no resonance in the pass band. This may be achieved by an ideal transmission line or a sealed enclosure operating below the fundamental resonance of the system.

Filter Analogies

A woofer is a mechanical high-pass filter. The design determines how steeply the response rolls off at the bottom and how much the filter “rings” (oscillates after the signal stops). Ringing in a woofer is often referred to as “overhang.” Overhang is an undesirable characteristic that muddies bass definition and makes the rhythm and pace of music more difficult to follow.

A sealed enclosure typically delivers a second-order high-pass characteristic. It rolls off at 12dB/octave with the least oscillation (overhang). A vented enclosure typically delivers a fourth-order high-pass characteristic. It rolls off twice as fast (24dB/octave) and oscillates twice as much after the signal stops. A passive radiator design is a vented enclosure with the passive radiator replacing the volume of air in the vent. They work the same way and perform pretty much the same except that a passive radiator can be weighted, allowing it to work in a much smaller box. Midrange frequencies, which might find their way into the room through a bass vent, will be blocked by a passive radiator.

A vented enclosure reduces driver excursion near the resonant frequency of the vent. This allows the bass/mid-driver in a full-range speaker to deliver reasonable midrange performance and go a little lower than it otherwise would. A vented sub-woofer will play louder than a sealed design while sacrificing bass extension and transient response (acceptable for home theater but not the best choice for music).

My Observations

I don’t like the sound of speakers that use vented bass loading. Transmission-line loading sounds a little better to me but not enough to justify the added cost and complexity. Sealed enclosures require equalization or enormous size for real bass extension. These are definitely potential drawbacks but EQ is the compromise that I choose to make in the real world.

These are generalizations and there are near-infinite possibilities for design compromises. Use my opinions as guidelines as you listen and make your own determinations about what sounds best to you. APJ

THIEL
CS2.4
Speaker System

Review by Shane Buettner

There are few companies in high-end audio with the well-deserved reputation for innovation and performance that Thiel Audio enjoys. Thiel has been manufacturing high performance loudspeakers in the U.S. since the late 1970s. The Thiel 2.4 is one of the company's latest designs, and is a relatively diminutive three-way floor-stander that sells for just $4,200/pair in satin black. (Thiel's typically lavish finishes can be applied for an additional charge.)

I'm very excited to be reporting on Thiel's 2.4 loudspeakers, which are truly high-end, high performance, time- and phase-correct and hand-crafted in the USA. In an industry riddled with over-priced and under-performing components, a speaker like this is a breath of fresh air.

While many magazine reviewers would have you believe that the carriage trade products they have on long-term loan are truly reference-quality, here's a speaker that many people can actually afford (without a second mortgage) that's demonstrably superior to those pretenders in a number of respects.

Regular readers also know that the **Journal** is a big proponent of time-domain fidelity in loudspeakers and that Thiel Audio is distinguished as one of the few remaining manufacturers of time- and phase-accurate loudspeakers. Since the demise of Dunlavy, Vandersteen and Meadowlark are the only others I'm aware of.

I share Richard Hardesty's opinion that time- and phase-correct speakers represent a higher standard of performance and a more sophisticated level of connoisseurship compared to conventional loudspeaker designs. For those who value convincing dimensionality in imaging and soundstaging I don't believe there's a substitute.

By their very nature, time- and phase-accurate speakers require more work and ingenuity on the part of the designer, and very high quality parts in the drivers and crossovers. It's no wonder most designers prefer to tell you that time-domain performance isn't audible—it's very difficult and expensive to properly execute a time- and phase-correct design.

If you're a new subscriber I heartily recommend Richard Hardesty's article *Time and Phase, Not Just a Craze* from the **Audio Perfectionist Journal** combined issues #6&7 for an outstanding primer on the importance of time-domain performance. That issue also contains an in-depth look at Thiel's design philosophies and an account of Richard Hardesty's visit to the Thiel facilities in Lexington, Kentucky.

Design and Construction

Many loudspeaker designs that sell for ludicrous sums are sold on the alleged integrity of construction in their cabinets and/or the rarity, quality and expense of the materials used in the cabinets, drivers and crossovers. After reading about the design techniques and materials used to create this $4,200/pair speaker from Thiel, I hope you'll cast the same jaundiced eye that I do toward the companies selling 7" two-way speakers (or a 7" two-way on top of a vented woofer box) for over twenty thousand dollars per pair based on spurious claims of construction/parts quality.

Cabinets

Perhaps the most unique appearance aspect of the Thiel 2.4 is the sloped and sculpted front baffle. The slope is used to maintain physical temporal alignment of the drivers, which, along with the coaxially mounted midrange/tweeter, obviates adjustment of the speaker's tilt in order to maintain optimal image

CS2.4

focus or tonal balance. The contoured edges of the baffle are designed to break up any early reflections from the drivers that could arrive at the listener's ears close enough in time to the primary signal to cause degrading time smear. Thiel believes the contoured baffle results in a more open and focused soundstage and my listening experience with the 2.4 backs up that assertion.

The 3"-thick MDF front baffle material is robust and rigid. The rest of the cabinet is 1"-thick MDF, and is braced to a truly extreme degree. The Thiel 2.4 cabinet is inert. Rapping the cabinet with your knuckles produces the dullest of thuds, with no sensation of resonance whatsoever.

I assisted Richard Hardesty a few years back when he updated the drivers in *Widescreen Review's* Thiel CS6 speakers to a newer model. I haven't seen any speaker at any price that surpasses Thiel CS6 structural integrity. Although the CS6 front baffle is made of a more advanced material, the 2.4 cabinet seems to be constructed to similarly high standards in spite of the price differential.

I had Thiel send the review speakers in satin black, but for an additional charge Thiel offers a number of exquisite veneers

that are mirror-matched on each speaker pair. You simply won't find finer craftsmanship or greater aesthetic appeal in a speaker cabinet no matter how much more you spend.

Drivers and Crossovers

The CS2.4 uses a newly developed coincident aluminum tweeter/midrange driver (1" dome tweeter, 3.5" midrange) that's

unique among coaxial drivers in that it uses a single voice coil—the so-called “unicoil” design. This coaxially mounted midrange/tweeter relies on a mechanical crossover (no electrical network) between the two drivers that Thiel claims conforms to a phase-coherent 6dB/octave acoustical slope.

As with any high quality speaker there is substantial cost tied up in a Thiel speaker in its drivers and crossovers. The unicoil system offers a very high performance-to-price ratio by using only one motor/magnet system and eliminating the electrical network. Jim Thiel told me that, while this driver is more expensive than a separate midrange and tweeter, it’s not as costly to implement as a typical dual motor midrange/tweeter, especially with the crossover eliminated. He essentially considers this design to offer “three-way performance at a two-way price.”

Coaxial drivers in general have advantages in a time- and phase-correct speaker. One is that the midrange and tweeter are temporally aligned by sharing the same space. A gentle slope of the cabinet is all that’s required to align the coincident driver’s output with that of the woofer, and the typically narrow vertical listening window of a time- and phase-correct speaker is expanded greatly. I’ve never found the narrow vertical window troublesome as my critical listening is done from the same chair at the same height every time. But some people may appreciate being freed from the perceived “head in a vice” constraint.

A potential drawback of using a coaxially mounted midrange/tweeter is whether the drivers stay effectively decoupled from one another and avoid intermodulation distortion. In other words, does movement from one driver cause unwanted movement in the other, which results in distortion? Thiel combats this by shaping shallow midrange cones to form a proper surround for the tweeter.

“…the midrange and tweeter are temporally aligned by sharing the same space…”

The 2.4 uses a single-layer aluminum midrange material in its unicoil driver, where Thiel’s more costly designs (such as the CS6 and CS7.2) use a three-layer sandwich material for increased rigidity and damping.

Using aluminum for the midrange driver material has tradeoffs. Aluminum is lightweight but very stiff and can operate over a broad range of frequencies with high resolution and low distortion, which is a prerequisite for drivers in a speaker with first-order crossovers. Aluminum drivers exhibit a so-called “oil can” resonance at certain frequencies, a malady Jim Thiel engineers around by making sure the resonances occur beyond the frequency range at which a given aluminum driver operates in his speakers.

Objectively, an impulse response test will show aluminum drivers as prone to ringing. For example, the driver resonates over a longer period of time after a transient than drivers made of softer materials. On the other hand, softer driver materials don’t operate in linear, pistonic fashion over as broad a range, and tend to absorb energy, either of which can result in lower resolution and compressed dynamics.

Subjectively, the most obvious drawback of aluminum in a midrange driver is that attentive listeners will hear the distinct sonic signature of the metal cone in this critical band. Aluminum midrange drivers have a sound that will appeal to some and turn others off. It’s my opinion that Thiel gets the most out of this design choice, absolutely minimizing, if not negating entirely, the potential pitfalls.

The 2.4 uses an 8” aluminum woofer with a 7.5” x 11” passive radiator. Vented/ported woofers resonate the mass of air in the port to increase the low frequency output of the driver. A passive radiator does the same thing by using the mass of the passive radiator instead of the mass of the air column of the

port but has the same time-domain characteristics as a vented box, which means more phase shift and group delay than a sealed box. When listening, that translates to bass that plays a little slower than the band is actually playing, which has obvious repercussions to rhythm and pace.

Passive radiators, however, are superior to vents in eliminating port chuffing (air moving through the port that’s annoyingly audible at the listening position), and eliminating any potential for the backwave from other drivers in the enclosure coming to the listener through the port too close in time to the primary signal.

Another unique design choice by Thiel is the use of “under-hung” voice coils—short voice coils in a long magnetic gap. A typical long voice coil/short gap motor system produces distortion in bass drivers because the power of the magnetic field acting on the coil and the amount of iron in the coil vary as the voice coil moves back and forth, toward and away from the magnet structure. With a short coil in a long gap, even when the coil has moved a long way, it’s still in a uniform magnetic field within the gap. Thiel further eliminates these distortions by using copper sleeves over the pole piece and copper shorting rings around the pole base to stabilize the magnetic field acting on the coil.

Thiel’s crossover networks are all hard-wired (no circuit boards) using the finest quality parts, including polypropylene and polystyrene capacitors, along with very pure, low oxygen copper, and air core inductors. The network in the 2.4 conforms to a first-order, 6dB/octave acoustic slope between the woofer and the coaxial midrange/tweeter. Thiel’s networks are complex as they compensate for impedance and phase deviations between the driver elements as well as damping driver resonances.

All in all, there’s enough engineering innovation and quality parts and construction in the Thiel 2.4 to flat-out embarrass many of the ultra expensive designs out there.

Adding Multichannel Capability to a Thiel Speaker System

Another consideration that may be enticing to some is found in Thiel’s unique options for adding multichannel based around their stereo speaker pairs. Their PowerPoint® surround speakers are attractive, versatile beyond belief (they can be easily installed on floors, ceilings or front, side or rear walls!) and constructed to Thiel’s exacting standards using high quality drivers similar to those in their floor standing designs.

Thiel’s MCS1 is designed for center channel use and employs a coaxial midrange/tweeter very similar to the one used in the 2.4. The long-awaited SmartSub® system is among the most innovative subwoofer/room correction system designs around, which I hope will be enough to one day lure Dr. Boom himself (Richard Hardesty) out of retirement from the field of reviewing subwoofers.

Setup and Reference System

I achieved the sound I liked best with the 2.4s just under 7 feet apart (center to center), with a distance of just over 9 feet from

each speaker to the listening position (an included angle between the speakers and listener of considerably less than 60 degrees, which is less than Thiel's user manual recommends).

In my 24.5' x 17.5' room the distance from the coaxial mid/tweeter to the respective side wall was just over 5 feet and each speaker was more than 5 feet from the front wall. This did not optimize bass performance in the room as there was little if any boundary reinforcement from this position, but produced the best soundstage with the most coherent, convincing, precise imaging and best soundstage depth.

The amplification components in my system during the Thiel review consisted of VTL's TL7.5 preamp, Theta Citadel monoblock power amplifiers (both of which I bought after reviewing them for the **Journal**), and Ayre's P-5xe phono stage. Source components included Ayre's D-1xe CD/DVD player, and a Linn LP12 turntable with all the latest accoutrements: Cirkus bearing, Lingo power supply, Ekos tonearm, Akiva cartridge, and the "Speed" carbon fiber mat from Extreme Phono.

My system cables are all AudioQuest's battery-biased lines: Sky and Cheetah interconnects and Kilimanjaro speaker cable. My power cords are AudioQuest NRG-5 and Richard Gray's Power Company High Tension Wire power cords. I also use RGPC 400S power line conditioners.

Listening

As soon as the Thiel 2.4s were set up in my room, they made great sound, immediately exhibiting the expansive soundstage and convincing image focus that sets time- and phase-coherent designs completely apart from conventional speakers.

Conventional speakers tend to create a strong image deep in the center of the soundstage that compresses sharply front to back at both sides, rather like a triangle. The Thiel 2.4s maintained layered image depth and excellent focus far out to both sides, and far behind the speakers. The imaging capability of this speaker can't be described as anything less than spectacular.

"The Thiel 2.4's soundstage was completely and utterly free of the physical boundaries of the speakers..."

The Thiel 2.4's soundstage was completely and utterly free of the physical boundaries of the speakers, and very open sounding. Not only did this speaker never sound boxy, it never sounded like it was there at all, even in the bass. Vandersteen's 3A Signature, which I owned for nearly three years, has better bass extension, but that comes at the price of sounding a tad boxy in the lower registers.

The Thiel 2.4 doesn't go as low, but you never hear the box either. I suspect this is a tribute not only to the cabinet con-

struction but also to Thiel's distortion-minimizing woofer designs, explained in detail earlier. Superior components like the Thiel 2.4 make your system sound less mechanical, less like a system and more like music naturally occurring in space. To that point, "free" and "open" are two words that pepper my notes on the listening sessions I spent with this speaker. Occasionally I felt this speaker gave up a rather teensy bit of ground to the dense, fully rounded 3-D imaging I get from my reference speakers. But take this nit-picking in its proper context. No conventional [phase incoherant] design I've heard at any price is even worthy of comparison to the spatial precision and dimension of the Thiel's soundstage.

Time is not the only domain in which the Thiels are coherent. Tonally the 2.4 is exceptionally balanced from top to bottom, as even and neutral a presentation as I've heard from a speaker. The top is airy and extended without being zingy or calling undue attention to itself, and the midrange is resolved, if a little cool (more on that in a minute). The mid-bass (50Hz-100Hz) sounds noticeably quicker and cleaner than typical vented designs, if just a tad over-damped. Combined with the aluminum midrange, this gives the 2.4 the cool signature Thiels are known for.

Low bass (50Hz and below) is where I'd describe this speaker having a slight subtractive coloration. Although its -3dB point is specified at 33Hz, low bass lacked size and impact in my room, but also sounded just a bit loose at the same time.

Big acoustic bass sounds, for example, were a little thin on body sound and the strings didn't snap quite as tautly as I'm used to. I'm certain I could have improved the bass extension by moving the speakers closer to the room boundaries, but my biases are such that I'd prefer to maintain the spatial performance derived from having the speakers out in the room.

If you want more bass from this speaker it's my opinion that you should add a quality subwoofer to your system and leave the speakers out in the room where they image best.

What about that aluminum midrange driver? Does it have a sound? Yes, in my opinion, it does. The Thiel 2.4 unequivocally does NOT have anything resembling the harsh, metallic sound of other speakers I've heard using aluminum midranges. (Monitor Audio and RBH are two examples of speakers I've heard using aluminum midranges that sound just plain nasty.) There is a damped, restrained coolness in these Thiels, and the midrange of the 2.4 is certainly not as relaxed as other speakers that don't use aluminum midrange drivers, including the Vandersteens I own. But there is a big difference between sounding cool and sounding metallic, and in my system these Thiels never crossed that line.

On harder recordings there's definitely less forgiveness than some are used to hearing, but there was no bite or glare either and this sound didn't prevent me from getting deeply involved with the music and thoroughly enjoying this loudspeaker. So long as you stay away from components with a pushed, hard midrange you'll hear open, highly resolved, slightly-on-the-cool-side-of-neutral sound from these speakers.

Further expanding the 2.4's charm is a very engaging sense of transient speed, dynamics and lifelike snap. Drum kits had outstanding pop and excellent dynamic contrast. Micro changes in voice level or the intensity of plucked strings were clearly apparent.

While the 2.4s were in my system they pulled double-duty for home theater playback and did an excellent job. I used Theta's powerful Citadel amplifiers (400W per channel into 8 ohms), and was more than pleased with how the Thiels performed in my system even when driven at demanding levels.

"I think at $4,200 a pair this speaker is one of the finest values in audio."

In terms of overall resolution this speaker outclasses nearly everything I've heard at or near its price point, and is frankly superior to many speakers I've heard that cost multiples of its price. If pushed I'd admit that I believe the Vandersteen 3A Signature can take you farther into the recording space, and reveals more low level detail, but I also think a lot of listeners will gravitate to Thiel's clean, open and lively sound. And if form factor becomes an issue, the Thiel has scoreboard, although that comes at a price.

Another factor in the valuation of this speaker is what's happening with the dollar. It is not my intent to get jingoistic here, but current events dictate that buying speakers made right here in the US of A has never been a better deal. With the poor position of the dollar in the world economy, and the fact that importers and distributors have to add a good percentage to the price of their products to cover their expenses and make a profit themselves, the odds are you'll pay more and get less from an imported speaker.

Conclusion

The Thiel 2.4 is simply superb. It occupies a small footprint in-room for a full-range, floor standing loudspeaker and can be purchased with a gorgeous furniture-grade finish that makes it an attractive and practical speaker to share your living space with. I feel compelled to mention the price because I think at $4,200/pair this speaker is one of the finest values in audio, but I also feel that mentioning the price denigrates this speaker in some respect. The Thiel 2.4 is not a terrific speaker at this price—it's a terrific speaker in its own right, regardless of all other factors, and I thoroughly enjoyed listening to it.

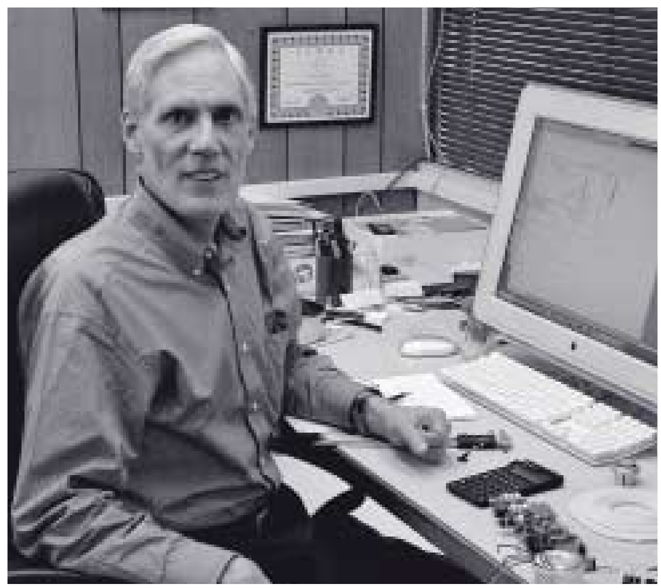
The Thiel 2.4 is a hand-crafted, high resolution, time- and phase-coherent speaker that will simply embarrass many conventional designs costing much, much more. That its looks match its high standard of performance is the icing on the cake. The Thiel 2.4 is the kind of product that high-end enthusiasts should celebrate! APJ

Thiel Audio
1026 Nandino Blvd., Lexington, KY 40511-1207
(859) 254 9427 www.thielaudio.com

An Interview with... JIM THIEL

by Richard Hardesty

I've known Jim Thiel (cofounder and chief designer) and Kathy Gornik (co-owner and company president) since the late '70s when I became a dealer for their speaker products. I sold,



installed and repaired Thiel speakers for many years and am quite familiar with the quality of their construction and the people who make them.

Jim, Kathy and I have maintained our personal friendship and Jim has been one of the truly knowledgeable engineers upon whom I have relied for education over the years. If I needed business advice I'd call Kathy and she was always ready to help. When I wanted to know about the function of an audio component I'd call Jim and he'd always share his experience graciously while supplying concise answers to my questions.

A few years ago my wife Paula and I visited the Thiel factory in *beautiful Lexington, Kentucky. This was an enjoyable experience that provided an indispensable glimpse into the sophistication of the current state of the art in speaker manufacturing. The value of Thiel products can best be realized when you see what goes into them before and during construction. Thiel speakers are among the most thoroughly engineered products available and an interview with Jim Thiel is sure to provide valuable information to* ***Journal*** *readers.*

Jim, tell us a little about yourself and how you got into the loudspeaker business.

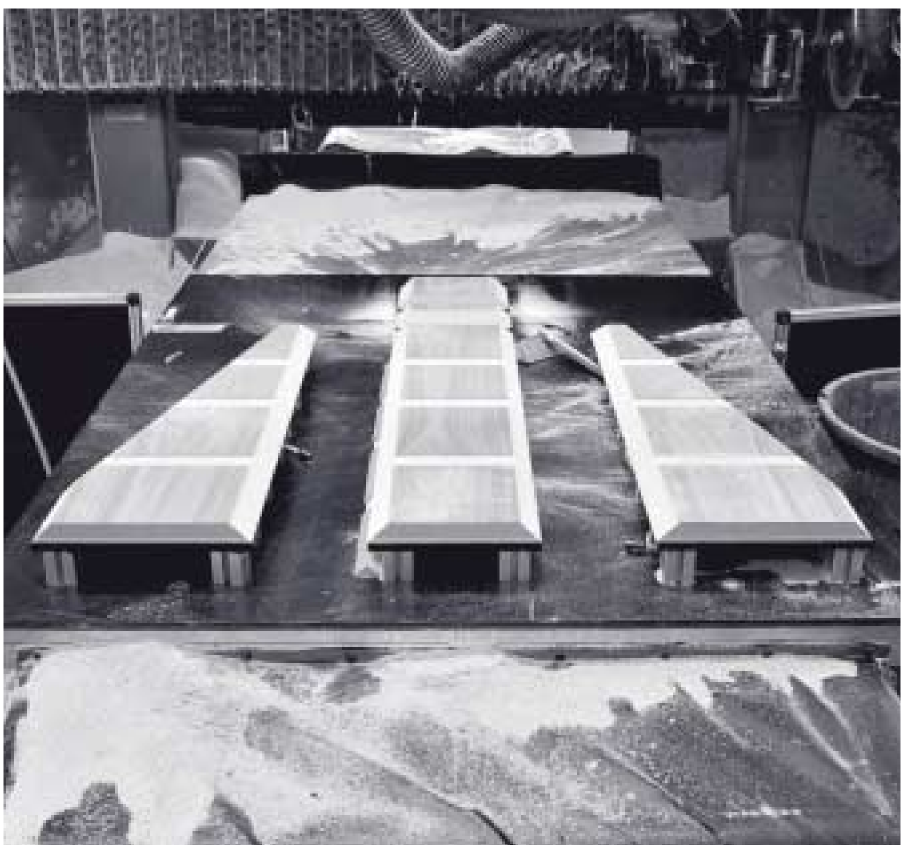
I've liked music a lot since I was young. I took piano lessons when I was 6, 7 and 8 years old. I played in my high school band. So that's part of it. I'm also a technical person who likes to work on challenging problems and great sound reproduction is, I think, a challenging problem. Nobody has succeeded in doing it perfectly, so we can always strive to get better results.

It's interesting to me to apply technical effort toward music reproduction and to think that when you're all finished what you really have is an audio product that's made out of metal and wood and plastic—and music comes out! And that's kind of magical. So this work suits my personality well and I really like it.

Had you been experimenting with making audio components for a long time, before you got into the business?

Yes, electronics was actually what I knew much better than acoustics and speakers and I had built amplifiers and preamplifiers and also band equipment, including guitar and PA amplifiers.

When I decided I wanted to start my own business I considered doing electronics like amplifiers and other electrical components. But I thought, rightly or wrongly, that there was more room for improvement in loudspeakers—and particularly more room to make improvements that people could appreciate. I thought that I could make a better amplifier but it might not be obvious, or appreciated by that many people, that it was a better amplifier. I thought that I might be able to make speakers that were enough better that a lot of people, just on hearing the product, would realize that it was better. So I decided to build my business around speakers, even though when I began I didn't really have any professional experience in loudspeaker design, other than having built my own speakers as a hobby through high school.



I've been happy with speakers because electronic engineering is involved, mechanics and acoustics are involved—a lot of different things, including material properties. Also, certain aspects of speaker designs don't readily lend themselves to measurements or pure simulations or solutions by mathematics alone. So you also have to bring a fair amount of intuition into the mix.

Time domain performance has been ignored by most of the major loudspeaker manufacturers and the magazines that review their products. Why are Thiel speakers time- and phase-accurate?

The approach I've taken to speaker design from the beginning is to consider it

a problem with solutions and my job is to find the solutions. Loudspeakers don't sound like a live musical performance. Why is it that when you close your eyes you can tell whether you're listening to the sound from a speaker or an actual musician?

I continually ask myself this question: what are loudspeakers doing that makes them sound like loudspeakers? And I've come up with a lot of ideas. For example, each note—from a piano or a guitar or whatever—is composed of a series of harmonics that, in real life, arrive at your ear at precisely the same time with precise phase relationships. A typical speaker takes that signal and breaks it down into different frequency bands, each of which gets to your ear at a different time with the phase relationships between them obliterated. So that is something that a speaker does to alter the signal and that would be one reason speakers don't sound like live musical performances.

I set up an experiment to test that theory. I built a loudspeaker that was phase- and time-coherent and did not have any of those distortions. I compared that speaker, when it was wired up to be time- and phase-coherent, with the same type of speaker wired up not to be phase- and time-coherent to see if I could tell the difference.

Here is, I think, one of the interesting things about loudspeakers. This was not a scientifically controlled experiment in the sense that the only difference between these two speakers was that one was time-coherent and the other was not, because it was not possible for me to make products that were identical except for that factor. So the test probably wouldn't convince a skeptical person beyond a shadow of a doubt that time-coherence makes an audible difference. But fortunately I don't have to convince everybody else—I only have to convince myself that it's worthwhile. Nobody has to agree with me, but if I'm going to make speakers I want to know for myself whether something is worthwhile. And I do think it's worthwhile and I can do it whether it's been objectively proven or not.

Anyway, listening to these two speakers, there was actually more of a difference than I expected, almost a jaw-dropping experience! The phase- and time-coherent speaker seemed much more realistic to me. There was much more sense of space and depth and clarity.

There were other tonal differences that probably could have thrown off an inexperienced listener, but when you have some experience with what tonal differences sound like you can hear through that and appreciate the difference that the other characteristics are making. So I became quite convinced that this was an important aspect of speaker performance—to make loudspeakers that sound more realistic.

Many of the diaphragms in Thiel speakers are made from aluminum. Why did you choose this material and what advantages does it provide?

because, although it's much stiffer and offers a wider bandwidth that's free of distortions produced by the diaphragm, at some high frequencies there are resonances that need to be compensated for or corrected. And some manufacturers aren't compensating or correcting these resonances, which create audible problems. These resonances can make the diaphragm sound like a metal diaphragm, which of course you don't want. So it becomes a little bit more involved to use a metal diaphragm effectively. The reason I consider it a better material is that we get drivers that perform well over a wider range of frequencies using aluminum diaphragms.

Ideally we want a driver diaphragm to be infinitely stiff and light enough so that we can maintain reasonably high efficiencies. The benefit of an infinitely stiff diaphragm is that it could move as one piece at all frequencies and not introduce any distortions or colorations that would result if it were internally resonating. We can't make a material that is infinitely stiff so we use a material that is as stiff as possible for it's weight and aluminum performs very well in that regard.

There are also other considerations and those have to do with the material being affordable, formable, and able to give very consistent results from unit to unit. Aluminum is very good in all these areas. You can easily form it into all kinds of shapes, it gives very consistent results and it's a very practical material.

The old paper diaphragms were really not bad at all. The wood fiber material that they're made of was in nature's R&D department for a billion years to evolve into strands with very high strength-to-weight ratios and they actually do a very good job. The biggest problems with the paper diaphragms had to do with inconsistencies from unit to unit, and from batch to batch.

Metal diaphragms are much more consistent but there's a complication with metal. You can't use aluminum indiscriminately

As the voice coil in a conventional drive element moves inward and outward it encounters a varying amount of ferrous material. This can produce variations in the inductance of the coil and is a mechanism for distortion. You use underhung voice coils in many drive units to eliminate this nonlinearity. Can you tell us why you chose this unusual construction method and how listeners benefit?

One of the reasons is exactly what you describe. Normally, as you said, the voice coil has varying amounts of iron inside the coil, depending on whether it's moving inward or outward while producing the sound and that changes the inductance of the coil and, therefore, changes the frequency response of the speaker.

So every time the cone moves in and out the frequency response is changing, as you described. By using the short coil and a long magnetic gap instead, all of the coil is always within the gap and, therefore, the amount of iron in the coil does not change and the amount of inductance doesn't change and, therefore, the frequency response does not change. But that's actually what I consider the second most important reason.

The larger benefit is that, because the short coil is always entirely within the magnetic gap, it always experiences a non-changing magnetic field strength. Therefore, it's able to produce a force that is proportional to the input signal from the amplifier.

Normally a voice coil is longer than the magnetic gap and, therefore, as it moves in and out, the amount of magnetic field acting on it changes. And that's the major distortion-producing mechanism in a loudspeaker. Usually, 90% of all the distortion produced in a loudspeaker comes from that mechanism within the driver motor systems. We can reduce that to a tenth or less of what it would otherwise be by using the short coil and a long gap. The only problem is that then you need a much larger magnet than you otherwise would and you also need bigger front plates and the cost of the driver is higher, but the distortion is much less so you get a much cleaner, purer sound.

I didn't invent this. It's like phase- and time-coherence that have existed in text books for many decades. Often, engineers don't choose to execute things because of cost or engineering difficulty. And I think there's some cynicism too, that nobody will hear the difference anyway.

In the early days of my business, I used to wonder, Should I be spending all this money and time doing this? Will anybody hear it? My answer to myself was, If nobody hears it and nobody cares, that doesn't mean I want to design speakers the normal way anyway. It really means that I would want to find another line of work. If I'm going to design speakers I want to try to design them well.

You want to design speakers that you can take personal pride in.

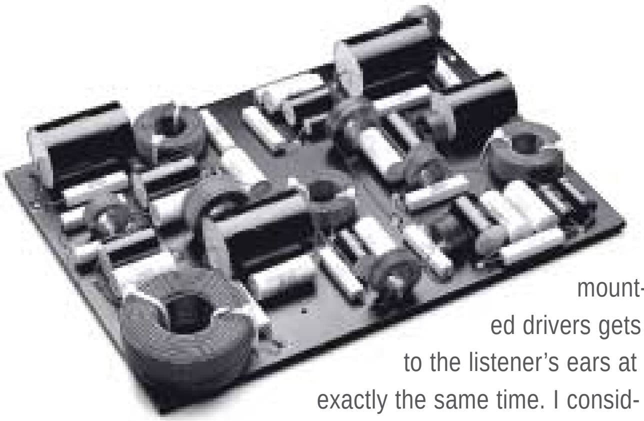
Exactly!

That's how everything good gets made, isn't it?

I guess so. But it's a pleasant surprise that it turns out there are people out there that can hear the difference—maybe not "most" people, maybe not a large enough group to allow me to buy a great mansion. But there are people who appreciate it a lot. We get letters all the time from customers telling us they've gotten so much enjoyment from our speakers, even the ones they bought 15 years ago, and thanking us for doing what we did. And that's pretty neat!

Many Thiel speakers use coaxially mounted midrange and tweeter driver units. Why did you go to the trouble of developing the special drive elements required to accomplish this physical arrangement?

That way, we can insure that the sound from those coaxially



mounted drivers gets to the listener's ears at exactly the same time. I consider that important, as we talked about earlier. The other methods (positioning the drivers on a sloped baffle) that we originally used, and still use, to achieve that time-coherence work very well in most cases, but not necessarily in all cases. A slopped baffle puts a limitation on the speakers that they can't be mounted on the ceiling or put up on a shelf because then you lose control over how far away the listener is going to be from each of the drivers. By mounting them coaxially, we can insure that no matter where the speaker is placed and no matter where the listener is, he will always hear the sound from those two drivers at exactly the same time.

Like a lot of things in loudspeaker design, solving one problem can create other problems. In this case, we found that you can't simply mount a tweeter in the center of a woofer because that affects the response of both drivers. Even though you can do that fairly easily to get correct time-coherence, other problems are created. For example, the tweeter really does not like being in the acoustic environment of the throat of a horn, which is the shape of a normal woofer cone. Even if you have a perfect tweeter, when you mount it in the center of a deep cone woofer its response gets screwed up. So we had to do something about that problem before we could use the coaxial mounting method. With attention to other factors, it does give you a great and precisely accurate way of achieving time-coherence.

Some of your coaxial elements have separate voice coils and use conventional crossover networks, and some use a single voice coil and a mechanical crossover to drive both diaphragms. Can you explain this for us?

As you said, many of our coaxial drivers are more or less conventional in that there is a self-contained tweeter with its own motor system that is mounted within the self-contained midrange with its own motor system. But we do make some

coaxial drive units that share the same motor system and use a mechanical crossover. There's only one magnet system and one voice coil, but that voice coil is connected to a mechanical crossover system that allows only the lower frequencies to reach the midrange diaphragm, whereas all of the frequencies go to the tweeter diaphragm, including the very high frequencies.

So that compliant ring that separates the two decouples them at higher frequencies?

That's exactly right. The coupling suspension, as we call it, decouples the midrange diaphragm at higher frequencies so it stops moving, leaving only the tweeter to move exactly the way a tweeter would normally move. The benefit of this system is that the costs for three-way speaker performance are really not much more than they'd be for a normal two-way system where you have just a woofer and a tweeter.

Here, we have a woofer and this mechanically coupled coax, but there is no additional electrical crossover network between the midrange and the tweeter and there's no third magnet system required for a midrange driver so the costs are not much higher than they would be for a two-way system, but you get the performance of a three-way system because the tweeter doesn't have to come all the way down to meet the woofer. You have a larger diaphragm than the tweeter operating in the midrange so I consider it a way to get much better performance for a small price than you can with a normal two-way speaker system.

I'm really happy about how this has worked out. This is an example, by the way, of one of the ideas we use that I did think was original but on investigation it turns out that this concept also was patented and used back in the 1930s.

Really? I'd never encountered that before. I've seen "wizzer" cones used in attempts to extend high frequency response of

CS7.2

midranges but I've never seen one with a mechanically decoupling crossover… I looked at that and thought, Gee, if he tells me how this works he'll probably have to kill me! This has got to be secret.

No, it's not secret. The difficulty is in making it work in practice. We literally had to make over 100 experimental drivers before we worked out the exact material and geometry needed for this coupling system in order to make the thing work well so it was a long project. It's kind of simple in concept but to have it work properly in practice was not so easy.

Many of your front baffles are sloped to temporally align the drivers and they are exceptionally thick and contoured to minimize stored and diffracted energy. Can you elaborate on how you arrived at your baffle configurations?

We've pretty much covered the high points. The baffle has several tasks to perform. The basic one is to hold the drivers in the position that you want. In our case, the baffles are sloped so we can hold them in the positions that cause the sound to reach the listener at the same time. Another task the baffle needs to perform is to not reflect or diffract any of the energy from the drivers so the driver is putting out its energy into the room unaltered by the edges or steps or angles in the baffle itself. That's the reason we round the edges of our baffles—to achieve as little interference as possible with the sound of the drivers.

The baffle, and the rest of the cabinet, should not generate any sound of its own and the only way to accomplish that is for it to be inert and not vibrating at all. The best loudspeaker enclosure would be made with thick, reinforced concrete or something that's extremely strong and won't vibrate at all—but that's impractical. Although we did use concrete in some of our big speakers. We approach this problem in the more moderately priced products by using very thick materials in our baffles. Depending on the model, our baffle material is 2 inches or 3 inches or 4 inches thick to minimize vibration. So if the baffle holds the drivers in position, does not reflect or diffract or interfere with the energy radiated by the drivers into the room, and does not vibrate itself then we consider that it's doing its job very well.

Besides the beautiful finishes what other special characteristics are incorporated into Thiel cabinets?

Stressing what we just talked about, ideally the whole cabinet needs to be so rigid that it will not vibrate. You have these drivers that are producing sound by vibrating in and out and sound is radiated from the front of the diaphragm and that's what we hear. But sound is also radiated from the rear of the diaphragm and goes into the speaker cabinet. The job of the enclosure is to completely contain that energy so it doesn't get out into the room to distort the sound that you're hearing—and that becomes very difficult.

Like I said, if we had foot-thick concrete cabinet walls it would be a lot better but that isn't practical. We use 1 inch-thick material for our cabinet walls and a lot of bracing inside the cabinet to reinforce the strength of the cabinet walls so that they vibrate much less and radiate much less distortion to the listener.

Your speakers are time- and phase-accurate so I assume that they use first-order acoustic slopes to integrate drive units. What other special design features do your crossovers employ?

Your assumption is correct; we do use first-order acoustic slopes to integrate the drive units. As you implied, that is the only way to achieve true time- and phase-accuracy. First-order crossover systems achieve not only accurate frequency response but also completely accurate time response and phase response and energy response. So they're completely accurate in every way and it's the only type of crossover that does have those characteristics.

"What the crossover network has to do is whatever is required so that when you add its response to the response of the driver the net result will be a first-order acoustic roll-off."

You're also correct in pointing out that it's the acoustic slopes that are important, and not the electrical characteristics of the crossover. If you were a great enough engineer to develop drivers that had first-order acoustic roll-offs in themselves, then you wouldn't need an electrical crossover network except, possibly, to keep low frequencies from passing to the tweeter.

So designing a first-order crossover is much more complicated than designing an electrical first-order network.

What the crossover network has to do is whatever is required so that when you add its response to the response of the driver the net result will be a first-order acoustic roll-off. The crossover network needs to work with the driver so you can't generalize about what such a crossover needs to do because the response that it needs to provide is different depending on which drivers it's working with.

In addition to the general requirement of working with the driver's response to produce the first-order roll-off characteristics, the crossover also has some other things that it must do. It needs to correct the response irregularities of the drivers themselves. And it must correct for alterations in response that the cabinet contributes. And, of course, it must precisely match the levels of all the drivers.

Another thing that we do, that usually isn't done, is to include in the crossover network additional compensation for the impedance changes of the speaker. That allows the speaker to present a much more uniform and resistive load to the amplifier. Now, technically that's not changing the sound of the speaker at all because these circuits are not in the signal path. But creating a consistent load allows some amplifiers to sound better than they otherwise would. It really depends on the amplifier

but some sound better—sometimes significantly better—by working into a resistive load rather than a reactive load.

So a crossover network has a lot to do and I think it's the most important element of the speaker. All the basic elements of a loudspeaker—the drivers, the enclosure, the crossover network—are very important, obviously. But I think the crossover network is perhaps the most critical.

*I believe that speakers should be demonstrably accurate **and** provide good sound. What measurements do you find the most informative? Do you measure in any unusual ways?*

Measurements are absolutely necessary but are not sufficient to evaluate loudspeaker performance. The one I consider to be most valuable—and the oldest one probably—is frequency response. I think that the only frequency response measurement that's of any value is a true anechoic frequency response. To my way of thinking, it's of no value to measure the so-called "in room" response of the speaker because you're really just measuring the response to the room and…well, that's another story.

I think you really need to know what the response of the speaker is independent of the room and, therefore, your frequency response measurements should be anechoic measurements.

"For lower frequencies I need to measure outside."

Unlike 20 or 30 years ago, the equipment now is low in cost because we use computers to achieve anechoic measurements without actually building an anechoic environment. But we still need the same size and space that was needed with the old anechoic chambers in order to get resolution down to workable low frequencies. So even today, you can't make useful anechoic frequency response measurements of a speaker at low frequencies in a normal room. For example, the room that I use has a 20 foot-high ceiling and the speaker is suspended halfway up and even that only allows me to get accurate readings down to about 200 cycles.

For lower frequencies I need to measure outside. It's tricky to get such measurements but the frequency response measure-

ments are by far the most useful. There are some caveats about that. What would look good on paper might not sound like a good frequency response measurement.

Your ear is so fantastically sensitive that, I believe, you can hear frequency response deviations of as little as one tenth of a decibel if they're across a whole octave. It's practically impossible to take measurements that accurately, and even if you could there are all kinds of things going on in the performance of the speaker that result in errors greater than .1dB.

Even a great speaker will have frequency response errors on the order of ±1dB, which is audible. But complications arise because it depends on what causes those irregularities. If you have a minor diffraction mechanism that's causing a response irregularity of 1dB, I've found that it's usually not very noticeable. But if you have a high-Q resonance that's causing a response irregularity of 1dB it can be quite audible and irritating. So you can't just say that if the speaker measures flat within ±2dB it's good—and if it doesn't it's bad. One speaker could measure flat within ±2dB and not sound good at all, and another might not measure any better in terms of frequency response but sound quite good.

It would be a boring world if we could measure everything, wouldn't it?

Yes, and we wouldn't be having this conversation and all speakers would be perfect and we'd be doing something else. But that's not the way it is. There are other measurements that can be taken—but I find that I don't take very many of the time response measurements or even phase measurements because all of that can be either calculated or is connected with the frequency response anyway.

Especially if you use an impulse as a stimulus...

Yes, that's a good excitation signal so that you can theoretically get all the information that is to be gotten in every domain from that signal. The real question about measurements becomes not which measurements to use—and you said this almost in these same words a few minutes ago—I think great measured performance is absolutely necessary but it is in no way sufficient. It's a starting point.

*The **Audio Perfectionist Journal** is primarily about accurate music reproduction but many of our readers also watch movies. Thiel makes a variety of home theater products. Do you do anything differently when you design a speaker for film sound?*

Well, yes and no. In terms of the basic performance values of accuracy, time-coherence, low distortion and wide dispersion and on and on, we don't do anything different at all and the reason, of course, is that an accurate speaker will reproduce all signals accurately and it doesn't make any difference to the speaker if the signal comes from a music source or a movie source. So a speaker that is accurate enough to be great for musical enjoyment will also be perfectly accurate for movies. So, in those ways we don't do anything differently. However, there are some practical differences and most of them have to do with how loud the speaker can play.

A lot of people want the speakers to play excruciatingly loud with movies. So something had to be done about this because speakers we traditionally make for listening to music won't play that loud. So what can be done is to make a speaker that's not nearly as extended in bass response. That doesn't directly compromise any musical values in terms of accuracy but it limits the range that the speaker is trying to reproduce.
That works well for movie systems if you have a good subwoofer in the system that's integrated well. So you could make speakers that would be good for movie systems that have a

"A lot of people want the speakers to play excruciatingly loud with movies."

subwoofer and would not necessarily be particularly applicable to a music system that didn't have a subwoofer. So most of the differences have to do with how loud they'll play and how low they'll go and if there's a subwoofer or not.

Tell us about your subwoofers.

I've been working on subwoofers a lot longer than I planned! It's been five years now and we finally have several products on the market.

The idea of great deep bass has always been very appealing to audiophiles. That's one of the things we love to have in our music reproduction. Unfortunately, great deep bass is difficult; it tends to make the speakers large and inefficient and expensive. Subwoofers came along and now we think we can have great deep bass. But in musical terms, most of them don't really sound very good at all. And they can sound kind of horrible! So many audiophiles think that a good music system shouldn't include subwoofers because they don't sound good. So what we've tried to do is to develop subwoofers that sound really great in musical terms—and I think we've succeeded.

There are a couple of problems that subwoofers have that other speakers don't have. Probably the simplest issue is that subwoofers are usually placed against walls and in corners and those placement positions really mess up the bass response of any speaker—not just of subwoofers.

You'd never take your great two-channel full-range speaker and put it back into a corner because, in addition to the other problems that would cause, it would really screw up the bass response. But that's where people often put subwoofers and when they get screwed up bass response they wonder why.

So we've developed a really nice way of correcting that problem right at the source, right at the subwoofer. You tell it where it is positioned in the room, how far away from the side wall it is and how far away from the front wall it is. Then it can know what effect that placement will have on its response and precorrect its response internally so that what it puts out in that position will give you the sound that would've been produced if it weren't near any walls.

And you do that with that "smart" crossover.

This part of it we actually do inside the subwoofer itself where you tell the subwoofer on the back panel what the side wall distance is and what the distance is to the wall behind the subwoofer. One reason this is done inside the subwoofer is that you may have multiple subs in different locations and may need different compensations. Another reason is that you might not have our smart crossover.

We've developed a new type of audio equipment. We call it the SmartSub® integrator. It's similar to an electronic crossover and its purpose is to blend the subwoofer with the main speakers and that's the other big problem. Even if subwoofers do produce high quality bass, which is a big if, their output is usually not well blended with the main speakers and what you end up with is a very separate sounding, disconnected bass with a different character, which sounds unnatural compared with the midrange.

So with the SmartSub integrator, you tell the unit about the characteristics of the main speakers and then it calculates the shapes of the crossover filters that will give you perfect results with that speaker. All the settings—what you want the low-pass frequency to be, what you want the phase characteristics to be—are automatically calculated for you to make the subwoofer blend with your main speakers, which it knows about because you have told it.

That sounds like a very complicated device.

It was complicated to design and yet it's very simple to explain. And I can tell you about it because we have a patent on it.

S+M=1 and so 1–M=S. So that's what this unit does. It takes the input signal, 1, calculates what the main speaker is going to be contributing, M, and subtracts that from the input. The result is the subwoofer output, S.

This allows the addition of a subwoofer—which can provide deep and powerful bass that is up to the standards of the most finicky audience, in my opinion.

Thank you Jim for an informative and enlightening interview!

Richard I really appreciate your interest and willingness to do this. I think what you're doing is great—not only for your readers and the industry but for me and our company. APJ

An Interview with...
RICHARD VANDERSTEEN *by Richard Hardesty*

In the nearly thirty years I have known Richard Vandersteen we have become close personal friends. My retail store was one of the very first dealers for Vandersteen speakers and I have first-hand knowledge about the engineering behind each of the

products in the line. I sold, installed and repaired Vandersteen speakers and offered my opinions about design and component choices for decades. My prejudices lean toward Vandersteen speakers because they embody the engineering choices that I have found to be important.

Richard and I don't see each other frequently but we communicate by phone several times a week and share ideas and observations about audio components and systems as well as a myriad of other subjects. We certainly don't agree on everything but we are in close agreement about how live music *sounds and about which audio components best replicate recordings of this sound. I believe that each of us has helped to mold the opinions of the other and I hold his engineering prowess in the highest esteem. He answers my questions about audio engineering and I tell him what I hear when these engineering concepts are applied to audio components.*

Richard Vandersteen and I have arrived at many mutually agreed upon opinions. We have reached these conclusions individually and together. We are predisposed to the ideas which we believe to be correct yet eager to adopt new positions when they are proven to provide a better explanation of the reality we observe. I own Vandersteen speakers.

Richard tell us about your personal background.

My parents came to the United States from Holland, the Netherlands. I was born into an immigrant family here in Hanford, California [between Visalia and Fresno], and like many families from Europe and Scandinavia we had no television set but there was always music in our home. My sisters played piano and my father was an accomplished singer. Music was always very important to us.

We had a console with two turntables, one of which was capable of recording by actually cutting lacquer discs, which we used to communicate with the family in Holland. We sent these discs back and forth by mail so that the family could hear the kids and also for basic communication. This device couldn't provide high resolution but it was an excellent introduction into recording technology.

In my youth I started experimenting trying to improve the sound of the family audio system—which of course was mono in those days—on a budget. I built some amplifiers and then some speakers, which I was continually improving as time went on. In the Air Force I studied electronics and after my stint in the military, I continued experimenting with loudspeakers as a hobby. Eventually that hobby became a business called Vandersteen Audio.

Isn't it amazing that vinyl discs are still the highest resolution source we have!

I'd agree, but I don't know that there's anything amazing about it. It's a simple system; it has more resolution than anything else we typically play in the home and I think it's charming and there's a little bit of a ritual to it but that's part of the enjoyment.

How did you actually get into the speaker business?

Havens & Hardesty, in southern California, and California Audio Systems, in Visalia, were the first dealers to place orders for the early Vandersteen products. I was reluctantly dragged to the Consumer Electronics Show in Chicago in 1977…

…Where you displayed speakers with response measurements that I made …

Yeah, and we ended up with a bunch of dealers and a bunch of orders—and that's what I've been doing ever since… Almost 30 years now!

Many manufacturers would have us believe that loudspeaker time domain performance is inaudible. Why are your speakers time- and phase-correct?

The very first speakers that I experimented with weren't time- and phase-accurate. I started by making speakers that provided different "flavors" of sound. At that time there was the "east coast sound" and the "west coast sound" and few talked about accurate sound.

When I considered building speakers professionally I gathered together a group of friends with high-end audio experience and asked which sound would be the most commercially successful. There was no unanimous agreement. After a few beers, I said, "Now wait a minute. What does the signal really sound like? What is coming from the amplifier? Shouldn't we replicate that as best we can?"

Accurately reproducing the signal from the amplifier became the new standard for this group. We would have long listening sessions in which we'd compare real sounds to recorded sounds. This predates our early experimentation with FFT computer analysis of speaker performance and we had to gather information empirically—by listening.

Using a Revox tape machine we'd record all kinds of things: various musical instruments, including piano, shovels scraping on the concrete, piling cardboard boxes, keys jingling on a key-chain, just all different kinds of noises and different sounds, trying to determine which of these speaker designs most accurately replicated the actual sound. We could determine that the signal passed through the playback chain including the amplifier with low distortion. All that was left was the loudspeakers and we'd listen to what came out of those and compare it to the actual sound.

During that process, we came to some conclusions about what was required for accurate reproduction. We decided that speakers needed only one driver per frequency range, first-order filters, minimum sized baffles and flat frequency response.

"We decided that speakers needed only one driver per frequency range"

The original speaker models had multiple drivers and large baffles with frequency response tailored to deliver a certain "sound." We ended up removing redundant drivers and the baffles in order to get rid of reflections from the box and reproduced sound became more like the source. Flat frequency response was necessary in order to accurately reproduce a variety of recordings. First-order filters unquestionably provided better replication than filters with steeper slopes.

And this just evolved over time until the very first commercial model that we made was time- and phase-correct, with first-order filters and temporally aligned drivers mounted on minimum-sized baffles. In the opinion of this listening panel, that's what it took to most accurately replicate these different sounds that we'd recorded.

So, your speakers are time- and phase-correct because you think that sounds better.

Whether something sounds better or not is a matter of opinion. What I was trying to do was to replicate the sound we'd recorded when playing it back through speakers right after having recorded it.

Why don't more manufacturers make time- and phase-accurate speakers?

It's a very, very tedious process. It requires high quality drivers, most of which must be custom-designed in order to provide a wide linear operating range. You cannot just design one and then duplicate it en masse or buy an off-the-shelf product.

It's important that the drive elements be reasonably flat in amplitude response and reasonably low in distortion and the crossover networks must be adjusted for each speaker to ensure that. So basically, what you're talking about is custom designing, in an anechoic chamber, a crossover for each and every loudspeaker made. That's very time-consuming and not really conducive to mass production.

Why do many of your products have simple cloth-wrapped enclosures?

The high labor quotient required for tuning and hand-making the crossover and the high quality drivers that are necessary—and the fact that we use more of them to produce four-way designs—leaves little money for fancy furniture, which really doesn't contribute to sound quality.

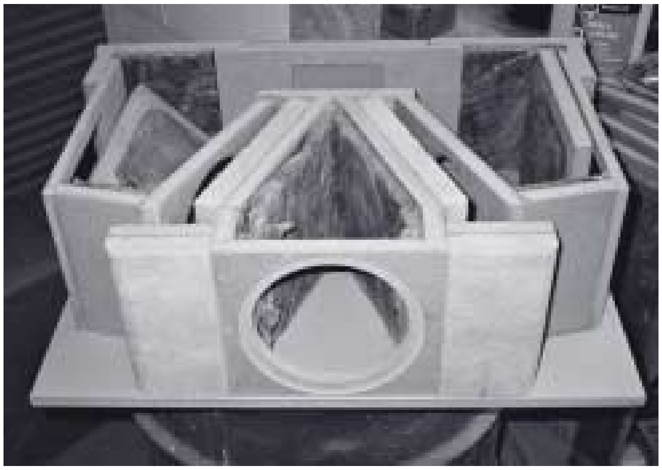
You can't really get satisfactory time- and phase-correct performance with only a two-way speaker. All of the criticisms that we hear about first-order filters are relevant only when there are too few drivers to share the spectrum. As the critics say, the IM distortion and power handling would be seriously compromised in a two-way speaker. So that means you're automatically relegated to using at least a three-way—we select four-ways in most models—and to divide the spectrum sufficiently so that each driver can handle the range in a linear way.

It's expensive to do that engineering correctly and to manufacture these products. And although our cabinets are sophisticated, they can be made with automated machinery and the finish is not important. So you end up with about 16% of the cost invested in the cabinets as opposed to 50-80%, which is typically the case with loudspeakers. Creating speakers with simple exteriors wasn't really a deliberate thing, but in order to do what we wanted to do at these different price points it was pure economics.

So, you're providing more performance for the money by not providing that fancy furniture.

Well look at it this way. First-order time- and phase-correct loudspeakers wouldn't otherwise have been possible because it couldn't have been successfully done. We couldn't have afforded enough high quality drivers and enough time to do the hand tuning of the crossovers if we had invested instead in cabinetry.

Why do Vandersteen speakers have a "family" sound?



Vandersteen Audio is all about preserving the waveform as passed to the loudspeaker from the amplifier. If speakers are reasonably flat in amplitude, and reasonably low in distortion, and if they are time- and phase-correct, and if they're replicating the waveform as accurately as possible then they're all going to have similar sound.

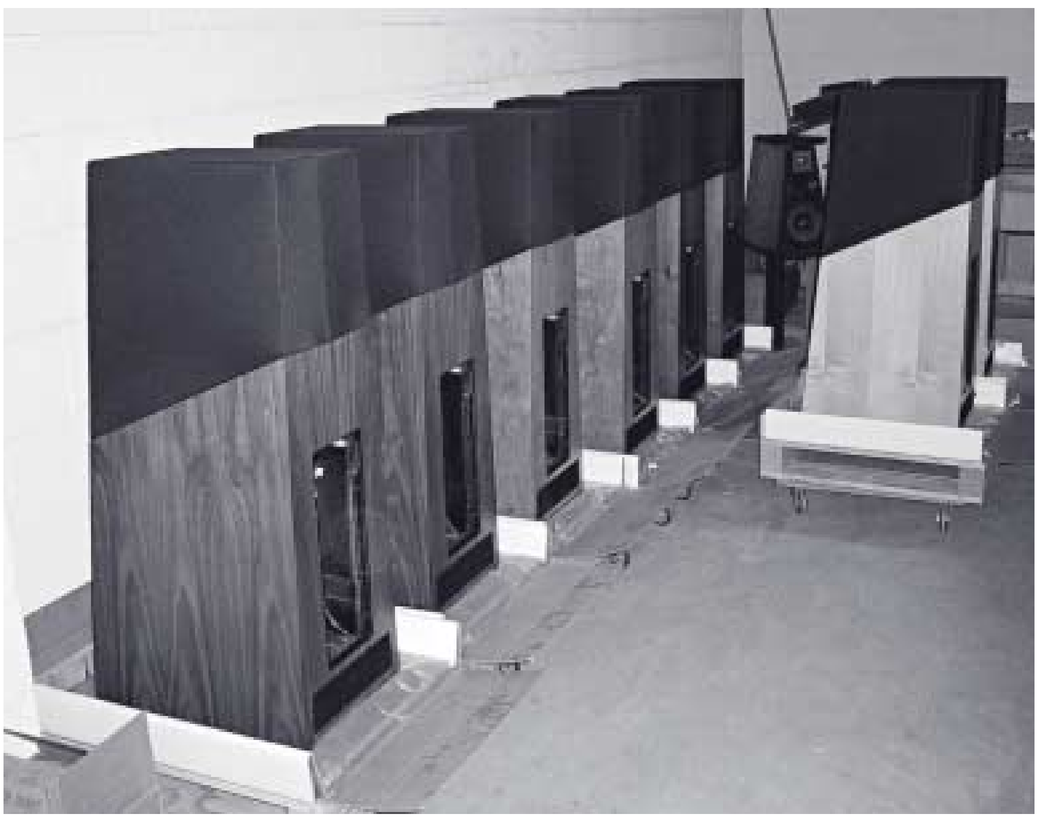
I've never understood these designers who create a line of speakers which all sound totally different from one another. They don't seem to have an actual opinion. All our speakers, on the other hand, are based on the same design principles because those are what I think are most important for reproducing live music, or reproducing music in a way that would remind you of hearing it live—in a piano bar, for example. So they should all sound very similar to one another. For more money, you're either getting lower distortion or better power handling or retrieval of more information—but the basic sound should be the same.

Many of your drive elements are uniquely constructed while other speaker manufacturers use off-the-shelf drivers or slightly modified OEM units. Why was this added expense necessary?

We do use slightly modified OEM units where that is adequate to the task because it's the most economical thing to do. The best examples would be woofers or subwoofer drivers. There are many commercially available woofers, for instance, which are perfectly adequate to operate up to 500 or 600Hz. Midranges and tweeters present a different situation. These drivers must be more sophisticated than what's available off-the-shelf. In many cases it became necessary to design special drive elements when no acceptable commercial units were available.

Our midrange unit, for instance, has a unique construction that eliminates reflections from the frame. That's a design I came up with and received a patent on. Because nothing like that was commercially available, we were forced to design and source our own components and have them assembled. The push-pull subwoofer driver used in the Model 5 is another example of a completely unique drive element. I designed it and made the tooling to manufacture it. It's actually assembled from my parts by another company that specializes in this type of work.

Your crossover networks appear complex but the signal path is often very simple. Can you explain this for us?



Because each filter is first-order the signal path is inherently simple This is the only crossover type that has the potential to be—but isn't necessarily—time- and phase-correct. However, for speakers using these filters to function acoustically all those extra crossover components are required to manipulate the driver's frequency and phase characteristics so that everything truly comes together at the listening position. The individual response characteristics of the driver and the crossover are less important than the acoustic signal that actually arrives at the listener's ear. All of our research considers the combined results at the normal listening distance.

So then, many of those components are to compensate for driver aberrations and aren't technically in the signal path.

None of the compensating components are in the signal path. All of them are in parallel circuits used to correct the frequency and phase response of the driver so that the speaker does what we want it to do at the listening position.

Doesn't that kind of debunk some of the naysayers who say, "Look at how complex these networks are"?

Well, the naysayers will always have something to say.

You pioneered the use of batteries to provide a constant bias voltage for capacitor dielectrics in crossover networks. Many people have called this a "snake-oil" approach. Is battery bias simply hype or does it really work?

Battery biasing was actually not my idea. Many years ago, before everyone became aware of how important it is to use film capacitors in the signal path, there was rampant use of electolytics. A certain listening panel applied a bias voltage to electrolytic capacitors and found that it made crossover networks using these capacitors sound a lot better.

...a properly designed PC board always sounds better than a hard-wired crossover...

I did some experimenting at the time and found that the same was true with film capacitors. Most of us have noted that there's a kind of warm-up that occurs in the first half-hour or hour of listening to even an excellent amplifier or an outstanding and well broken-in pair of loudspeakers. It is my opinion that the audible change that occurs in that first 30 minutes to an hour or so is due to dielectrics forming. The whole idea of battery biasing is to keep the capacitors in the same state they'd be in if they'd been played for 24-hours or so. There's nothing mystical about this and it certainly does work.

Speaking of dielectric materials, what capacitors are used in Vandersteen crossover networks? What resistors and inductors do you prefer?

We wind our own inductors; we use only air core inductors in the signal path. The capacitor dielectrics vary. They do sound different. We haven't found one particular dielectric to be a favorite in all instances; we often use combinations of three different ones for any capacitance that is in the signal path. We use styrene, polypropylene, and in some cases even Teflon™.

We use non-inductive resistors that are made without any steel parts and have copper leads and we've found that makes an audible difference. They're only available up to 5 watts so we often have to use multiple units.

Hard-wired crossovers are heavily advertised by some manufacturers and touted as superior. Why do you use supposedly inferior circuit boards?

Over the last 25 years I've found that a properly designed PC board always sounds better than the hard-wired crossover from which the board was designed because all the electrical characteristics are far more predictable and uniform. Exact component positioning and consistent electrical impedance assures uniformity and helps to ensure that the left and right speakers will be identical.

Stereo imaging is enhanced when the left and right speakers are exactly the same. Even with speakers that aren't time- and phase-correct, closely matched speakers will image better than those with even slight differences between left and right. We've recently released a new model called Quatro, for instance. We first brought the prototypes to the CES show two years ago. Everyone at this year's show commented about how much better the Quatros sound now. There was no design change; the only difference between these two products is the crossover, which went from point-to-point wiring on the prototypes to PC boards on the production units. They both have exactly the same drivers and design. Many of our dealers asked, "What did you change? It sounds so much better."

Vandersteen speakers have used minimum sized baffles to support drive elements since the beginning. How does this affect sound?

Speakers with large baffles are easier to make because the baffle reflections randomize and "smooth" the response from the drivers, especially in the crossover region. Time smear is the negative result of this smoothing.

Obviously we have baffles, but very small ones to hold the drivers. If you minimize the size of the baffles you no longer have this reradiation support and you have to achieve flat response by using better drivers and more closely matching the crossover networks. By eliminating this time smear, Vandersteen speakers achieve far better transient performance and provide more detail while remaining smooth.

What other special construction techniques are used in your enclosures?

Loudspeakers create a tremendous amount of energy and that can be a big problem. We do lots of different things at different price points to eliminate or compensate for the energy that vibrates the enclosure rather than producing sound.

It's very expensive to make an enclosure that's totally inert and impossible to completely achieve that goal. We attempt to make speakers that are acoustically inert by using different sizes and thicknesses of different materials. If there is some secondary radiation it's spread out so that there is less energy over a wider range of frequencies rather than more energy concentrated at a couple of frequencies. Also our Models 1, 2 and 3 are tuned to make sure that that residual energy is in an up-and-down (vertical) motion not a fore-and-aft (horizontal) motion, which could affect the performance of the drivers.

Do the dowels that support the cloth wraps on Models 2 and 3 have a negative sonic impact?

That's a good question and we've been criticized about that a lot over the years. I have proven by actual demonstration that the effect of the dowels is completely inaudible.

I love to give seminars because it puts me in touch with consumers. I can learn what's on their minds and how they're using the stuff; it makes me more aware of their situations. There's always a cynic in every group and at one particular seminar he just kept coming after me about the dowels. I borrowed a couple of brooms from the dealer. We blindfolded the cynic and I set one of the brooms right smack in front of the driver—not 120 degrees out to the right or left as the actual dowel supports are. The other speaker didn't have a broom.

We used a mono source and compared A versus B. The guy agreed that he couldn't distinguish between A and B and when we removed the blindfold he was quite shocked to see this broom handle, which is about the same diameter as the dowels in our loudspeakers, sitting right smack in front with no perceivable effect on the sound. Now if that is true, and we proved that it is, then moving those dowels far away from the natural radiation pattern should be even better.

Some Vandersteen speakers are specially designed for home theater use. How do these designs differ from your purist audio products?

Well, they don't really. They're all time- and phase-correct and have flat frequency response. If there is a need for accurate waveform replication it may be more noted by those who're into home theater because, unfortunately, most people are more familiar with the sound of car doors slamming and phones ringing than they are with live music.

They may not be familiar with what a piano sounds like without amplification in a little nightclub but they're certainly familiar with the everyday sounds that they hear in the movies. I know these sounds are done with foley effects and are not necessarily accurate but the foley artists are very good and they create sounds that are remarkably lifelike.

The first year we introduced our center channel at CES in Chicago, my wife and I and a sales manager from another company were in the back of our long, skinny room. On the right wall of the room was a counter with a phone above it. Over the three or four days of the show we were showing the film “Always” and during a tense control tower scene, a telephone rings in the tower (on screen). Everyone viewing the film in our room would look over to the right side wall and wonder who was going to answer the real phone. It’s an interesting effect and I think that replicating the waveform is every bit as important in home theater as it is in audio. To do home theater properly you need a very fine audio system.

“…most people are more familiar with the sound of car doors slamming and phones ringing than they are with live music.”

You have a unique center channel speaker, the VCC-5. Can you tell us about it?

The VCC-5 is a point source that uses four drivers. Because you don’t know where it’s going to be located it needs to sound the same no matter where you are, or it is, in the room—right, left, up or down. This was a real challenge.

There was demand from the dealers for a bigger center channel speaker that could play louder. Our small VCC-1 is a coaxial design, point-source radiation, time- and phase-correct center channel speaker. So we built a prototype of a larger center channel speaker carefully adding two woofers—one to the right, one to the left—and set the crossover very low at 100Hz in order to get more dynamic range and more low frequency reach, and also lower distortion, because that’s what a bigger center channel should do.

With these added drivers we encountered a comb filter effect and the speaker sounded different to each listener in the room. This is exactly what a center channel speaker is not supposed to do! A center channel speaker is supposed to anchor dialog

to the screen for listeners sitting off-axis. The VCC-1 does that. Our challenge was to make a center channel that played louder and went deeper but still worked as it was supposed to.
We needed to find a way to add two woofers in order to get higher levels with lower bass and lower distortion and to blend them physically with a coax to make the speaker sound as if it were a triax, so that all of the sound comes from the same point. This would provide a center channel speaker that was time- and phase-correct in all positions in the room—not just for the guy sitting on the crack of the couch but the person on the right and the person on the left and the person standing up or sitting on the floor.

All this work culminated in a patent-pending design that’s called the VCC-5. It took us a year to accomplish that and we’re very proud of it even if few people understand how it works.

Vandersteen makes some unique subwoofers. How do the 2Wq and V2W designs differ?

Well, the 2Wq is based on the 2W that’s been in the line now for about 20 years. Many things have changed over those decades and the V2W provides an answer for some of the new requirements that have arisen.

Twenty years ago, sub-woofers had a very bad reputation because people couldn’t get them to blend, they stuck out like a sore thumb and there was a lot of negativity about them. We spent many years perfecting a subwoofer that would blend seamlessly with the main speakers. We wanted to make it sound like your main speakers got bigger and gained authority. We found that the mids and highs were improved and imaging got better when the low frequencies were removed from the main speaker structure and from the amplifier driving those speakers. We found that the character of the main amplifier could be retained into the deep bass with a passive, first-order high-pass filter system that also helped provide a seamless blend.

We still get calls every now and then from customers who say, "I bought one of your 2Wq sub-woofers for $1,300 and I can't hear it." To me, that's a compliment not a complaint! But for some people, when they spend $1,300 they want to hear it, they want to know they got their money's worth. That's not what we were attempting to achieve with the 2Wq.

2Wq

The V2W, on the other hand, uses the same three drivers, slot-loaded and everything, but it is a little more traditional although it still has good transient behavior. It uses a large passive radiator to allow it to play substantially louder and accepts a line-level input. In home theater, people require and enjoy more energy. They don't quite want to hear it the way it really is. The V2W doesn't have the ability to mimic the amplifier driving it because it's driven line-level not speaker-level by the customer's amplifier. But it's got more punch.

In systems doing both music and home theater, some people are using one of each. The 2Wq and the V2W subwoofers look identical so they set a 2Wq in one corner so they can run their main speakers full range while they're listening to music and have the benefit of a subwoofer. In the other corner they put a V2W that is fed by the LFE output from their surround processor. When they're watching movies both the 2Wq and V2W are working and when they're playing music they're using the 2Wq only.

When you select "small" for the main speakers with surround processors you're introducing at least a second-order (if not higher) high-pass filter. If you have all Vandersteens you have a time- and phase-correct loudspeaker system but you do not have a time- and phase-correct signal chain.

The advantage of using the 2Wq for the main speakers is that it allows you to designate the main speakers as "large" thereby getting a first-order transfer function and avoiding that second- or higher-order filter that's in the processor. You can experiment with that just by switching between "large" and "small" and listening to what happens to the naturalness in the mids and highs.

The passive high-pass filter used for integration of the 2Wq subwoofer and the subwoofer sections of the Quatro and Model 5A speaker systems is claimed to offer some unique advantages. How does this system work and why is it better?

Audiophiles get very nervous when you put an extra component in the signal path because we know that any time you add an extra component there's going to be some degradation. This is simply a fact. Although the battery-biased high-pass, which now has a biased cable on its output, is very neutral and is made with extremely high quality parts, we know that it still has some negative impact.

However, high-passing the amplifier driving a loudspeaker makes such a stunning positive difference that it's like taking one step back and four steps forward. That's still a net gain of three steps.

Years ago, a company came out with a thing called a "warp knot" that was a device that plugged into your amplifier input. It was simply a subsonic filter to get rid of noise from record warps and other deep energy that you can't hear anyway, with a first-order transfer function. Even though it was made with a very low quality capacitor and it had a fairly significant sonic character, it was still deemed by many people to provide a positive sonic contribution.

Vandersteen high-pass filters perform a similar function at an even higher frequency allowing the amplifier to have an easier life. Even "arc welder" amplifiers sound better when they're not having to work hard at infrasonic frequencies. So do speakers.

If you can make a high-pass filter so transparent that the step backwards is very small, one has to focus on the fact that you've moved four steps forward in all other areas. Your amplifier and main speakers will sound better and you'll have better bass from the subwoofer(s) you've added.

To demonstrate these facts at seminars we've misadjusted the high-pass frequency to below 30Hz so the speakers were running pretty much full range. Then we've compared the high-passed speakers to speakers that were actually running full range. Virtually everybody has preferred the sound of the system with the high-pass filters in the chain because, even though they're hearing all the information that's on the recording, the subsonics and so forth have been attenuated. In all the times I've done that experiment, I've never had anyone prefer the system without the high-pass filters in the signal path.

Then what happens? After you've high-passed the main speakers, the woofers replace that information by...

The woofer sections in the 2Wq subwoofer, and the Quatro and the Model 5A speakers, apply the inverse of the roll-off created by the high-pass filter. Additional equalization is applied to adjust for the falling bass response of the sealed enclosures and the result is fed to an amplifier(s) specifically designed to drive the load of the driver(s) used.

Many people think that if you set a high-pass filter at 100Hz, 99Hz disappears. Actually it's only reduced slightly in amplitude and the signal level continues to fall as it goes lower in frequency at 6dB per octave. The entire signal is still there; its amplitude is simply reduced. It's like the RIAA curve for a phonograph record. That's why the amplifier likes it. And the main speakers like it.

All that low-frequency information is dramatically diminished from the amplifier circuits and the loudspeaker structure. The subwoofer amplifiers have the inverse of that built in, to restore flat frequency response. The only penalty is noise because we're building back on top of something that's been rolled off. However, with modern solid-state technology the noise floor is so low that that's not a factor.

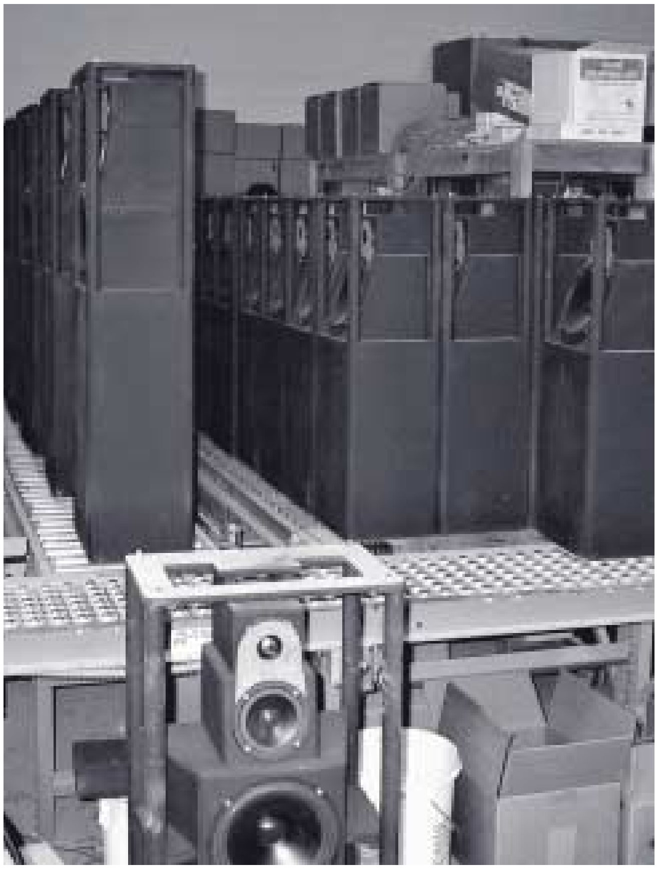
Tell us about the unusual subwoofer driver in the Model 5A speakers.


Model 5A

Around the factory here, it's commonly called the jackhammer. It's a completely unique driver that continues our commitment to reduced bass distortion.

We've always done proprietary things in our low-frequency drivers, which are required for long excursion. A woofer does not behave linearly on its inward stroke versus its outward stroke and that's partly because of nonlinearities in the suspension and partly because when you drive the voice coil inward there's more iron in the coil versus when you drive it outward there's less iron. In all of our speakers, we extend that iron out and add faraday rings to minimize the inductance change. Even with these additions some slight nonlinearities remain.

The push-pull subwoofer driver in the Model 5 has a sandwich cone attached to a voice coil former that goes all the way through it from one motor to another. The cone is shaped like a very thin flying saucer and there's a motor on both ends of the voice coil so it's truly driven push-pull and it's truly linear over its one and a quarter-inch stroke. Motion is always identical in positive and negative directions.



So all the nonlinearities would be cancelled because you've got everything duplicated and the two systems are operating in opposite directions?

Exactly. There are four amplifiers with a bridged pair driving each of two voice coils wound on the opposite ends of a single former that goes all the way through the cone. Each voice coil has its own motor system and magnet with the whole thing operating push-pull.

So everything is push-pull all the way, electrically and mechanically.

Right. The Model 5A and the Model 5 have four amplifiers in each speaker. There are two balanced bridged pairs, with one pair to drive each voice coil.

The Quatro uses two 8-inch drivers (the same ones that are used in the 2Wq) and two amplifiers that are paired into a fully balanced bridge to drive them. Using multiple small drivers, we're not trying to hit as high a peak level and linearity is not as much of a problem. Of course the Quatros are half the money of the 5As.

Thank you Richard. This has been an enlightening interview and I'm sure the information it contains will prove useful to our readers. APJ

The Engineering Double Standard

by Richard Hardesty

The magazines and the reviewers who influence the high-end audio industry have adopted a double standard for engineering competence that has had a negative impact on all of us. This double standard is pervasive. They expect all audio components other than speakers to meet certain minimum standards of objective performance. At the same time they treat speakers as if they were works of art to which no objective standards apply. This position is both false and misleading.

There are objective tests to gauge the fidelity of speakers and these tests are frequently more comprehensive than the recognized tests that gauge the fidelity of other components. Objective tests can't tell the entire story but they certainly can expose those products which are poorly engineered and can't possibly reproduce the recording accurately.

Amplification Components

Preamplifiers and amplifiers are expected to have wide bandwidth and low distortion and they certainly should. An amplification component with more than 1% total harmonic distortion is looked at with suspicion and a product with bandwidth that is limited to below 200kHz is often viewed critically. An amplification component with frequency response variations of ±2dB would be severely criticized.

I've read comments like "this product measures like it's broken" in reviews of amplification components but never in reviews of loudspeakers, many of which measure far worse!

Even digital switching amplifiers, which exhibit high frequencies that are audibly soft and "closed down," typically have bandwidth to 60kHz or above, which is three times the generally accepted limit of audibility. No one would recommend that you buy an amplifier that delivered the midrange frequencies out-of-phase from the rest of the spectrum and no amplifier that I've seen does this.

Magazine reviewers hold amplification components to strict standards of objective performance. Contrast these standards for performance with the rave reviews you'll read about speakers with frequency response variations of ±10dB, high frequency response that falls like a stone above 10kHz and midrange drivers wired out-of-phase with the woofer and tweeter.

Some magazines publish these speaker measurements, showing that the products are poorly engineered but the reviewers often love them anyway. Other magazines use these conflicts to "prove" that measurements don't mean anything. They're wrong because the measurements do mean something—the reviewers can't hear!

> ***"No one would recommend that you buy an amplifier that delivered the midrange frequencies out-of-phase from the rest of the spectrum."***

Disc Players

Digital disc players are expected to have flat frequency response and low jitter and these are necessary requisites for good performance. Jitter is measured in picoseconds, which are incredibly short periods of time. Do you think these errors will be audible on a loudspeaker that is totally incoherent in the time domain? If the high frequencies arrive at the listener several milliseconds before the fundamentals will a few picoseconds of jitter be detectable?

I recently read a review of an SACD player that was criticized for frequency response that didn't exceed 30kHz. In the same magazine there was a rave review of a loudspeaker system with a ribbon tweeter that was described as "the star of this show." It had virtually no measurable output above 12kHz where a sharp resonance appeared.

Another speaker system with an exorbitant price tag received page after page of accolades yet produced a frequency response graph that resembled a profile of the Grand Tetons. These measurements were described as "enigmatic." What's truly enigmatic is the response of the reviewer. Couldn't he hear these gross flaws?

Analog Source Components

Turntables are expected to turn at the correct speed within narrow limits and exhibit low levels of wow, flutter and rumble and these are reasonable standards for good audible performance. These factors are much more difficult to measure so you'll hear far more about picoseconds of jitter from CD players and signal-to-noise ratio misrepresented as dynamic range from compact discs.

Vinyl records clearly have greater dynamic range and wider bandwidth than compact discs and this fact is easily heard when the caparison is made using accurate loudspeakers. Could the popularity of the compact disc be partly due to factors other than convenience? When the magazines discarded the notion of high fidelity in loudspeakers did they doom a generation to a life of musical dissatisfaction?

Speakers

I think all high-end audio components should be held to high standards of objective performance. Loudspeakers are an important component in an audio system. As this **Journal** has explained, speakers—like all audio components—should have flat response with minimal energy storage and they should be time- and phase-accurate. It's time to hold speakers to the same objective standards as other audio components. Let's expose the pretenders even if it costs the magazines some advertising revenue. APJ

Paula T. Hardesty, Publisher
Richard L. Hardesty, Editor
Edith Hardesty, Copy Editor
Shane Beuttner, Equipment Review Editor
Rick Johnson, Art Director

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