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Richard Vandersteen of Vandersteen Audio: Part One

by Doug Blackburn (db@soundstage.com)

What's the best way to decompress after a five-day high-pressure event like HI-FI '98? Take a couple of days off then go on another audio safari involving six hours of driving! In this case, a factory tour of Vandersteen Audio, the first ever done, and an interview with the company founder and energizing force, Richard Vandersteen, were my goals. The factory tour will be coming up, but the interview was just too interesting (and too long!) to wait that long. So we'll be giving you the interview in digestible pieces to help you get through it all.

We don't often get to hear the deep-down rationale behind the design of high-end products, but in this interview, Richard Vandersteen makes some interesting observations about his speakers, other types of speakers, the criticisms Vandersteen loudspeakers have taken for 20 years from some quarters of the audiophile brotherhood, and other interesting facts both small and large.

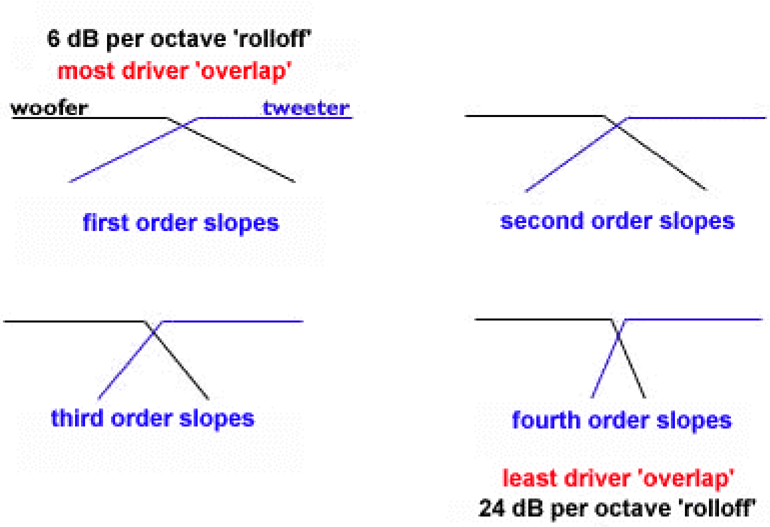
I started the "Vander-Day" from my brother's home in the western suburbs of San Francisco. Vandersteen Audio is located in Hanford, CA almost midway between San Francisco and Los Angeles. This meant a three-hour drive down the simultaneously fascinating and boring Interstate 5. It's fascinating because much of I-5 follows the western edge of the San Joaquin Valley where some unbelievably large portion of all the food grown in the US comes from. The fields are truly amazing. But the road is flat, straight and featureless. Where there is so much growing vegetation, there are *bugs* and the windshield was nearly covered with smashes within 60 to 90 minutes. This necessitated cleaning the windshield two times at gas stations during the three-hour drive.

Once at the Hanford exit, there was another 30-mile drive east toward the Sierra Nevada mountains which remained snow covered along the top 20% of their peaks. Arriving in Hanford, you are struck by the feeling of California the way it used to be -- before LA boomed, before Silicon Valley, before the crazy real estate markets. Vandersteen Audio occupies a well-kept industrial building just off the main east-west highway through Hanford. People and mail use the door on 4th Street. Shipping and receiving percolate through the dock on 5th Street. To get away from the factory noise and telephone interruptions, Richard and I talked at the Vandersteen homestead, less than a 15-minute drive from the factory.

We start our interview with a question about first-order crossovers, one of the hallmarks of Vandersteen loudspeaker designs. You can't help thinking that the additional effort needed to build successful first-order crossover loudspeakers wouldn't be expended if there weren't some awfully good reasons. Richard Vandersteen thinks he has some awfully good reasons for making speakers the way he does.

A Brief Explanation of First-, Second-, Third-, and Fourth-Order Crossover Slopes

In the figures below, the black lines represent the operating range of a woofer. The blue lines represent the operating range of the tweeter. You will notice as the crossover slope changes, the amount of overlap in the range of frequencies covered by the two drivers decreases. In the fourth-order crossover, there is a very limited amount of overlap between drivers. Normally, you would like to have a small amount of overlap between the two drivers; however, the electronic components required to create second-, third-, and fourth-order crossover slopes increase time delays and phase shifts that are applied unequally to the signals going to each driver.



NOTE: The graphic above is intentionally simplified. It is used only to illustrate the concept of crossover slopes. The graphic does not accurately depict frequency response of a loudspeaker.

The flat parts of these simulated response graphs indicate the main operating range of the woofer and tweeter. The sloped portions show how quickly the output of the woofer and tweeter drop off. The point where the woofer and tweeter response touch is supposed to be the -3dB point, meaning this is the point where the woofer and the tweeter are 3dB down in response from flat. Below the -3dB point is the zone where sounds are reproduced by both the woofer and tweeter simultaneously, the overlap zone.

Doug Blackburn: *Tell us about how you came to decide that you were going to build your loudspeakers with first-order crossovers.*

Richard Vandersteen: In the '70s we were beginning to play with loudspeaker designs and the realities of what external driver diffraction due to the enclosure surfaces and internal diffractions inside the loudspeaker enclosure were doing sonically. We started to look at things that would remove these secondary distortions that come from secondary radiation. Secondary radiation is most easily defined as sound which exists because of the reproduction method/device. Secondary radiation is 100% distortion because none of it exists in the original input signal. The more secondary radiation you can remove or prevent, the better the speaker will sound.

To see what was possible sonically without the secondary radiation products, we mounted drivers naked in space, no enclosure or baffle at all. In the '70s, second- and third-order crossovers were by far the most common types in use and that is what I was working with. Experimentally, but almost by accident, we discovered that a tweeter in this free-air fixture took on a completely different sonic character depending on whether it was driven by a first-, second-, or third-order filter. It's wasn't just different because the 3dB-down points were different for each crossover or because of the slower roll-off in the lower-order crossovers. The tweeter took on a noticeable pinched, twangy sound with a second-order crossover, which was even more noticeable with a third-order crossover. Only with the first-order crossover did the tweeter sound natural.

In this case, "natural" refers to how accurately the driver or drivers under test reproduced sounds and music we recorded. Besides musical instruments and voice, we would record things like a shovel scraping on concrete or shaking car keys. We'd then use the same shovel on the same bit of concrete in the test area, live, to compare how drivers (and crossovers) sounded playing the recordings of the same sounds. It was really helpful to be able to duplicate these recorded sounds live right in the same space where the drivers and crossovers were being tested. We tested woofers, midranges and tweeters separately and as sets to see how individual and ensemble performance of the drivers worked in these controlled conditions. In fact, one phase of testing involved putting the driver setup that was under test behind a curtain and bringing in a panel of listeners who could not see what they were hearing and did not know what they were listening to. We found from this testing that our own observations and the observations of the blind panel were in strong agreement: The setup with first-order crossovers always sounded far more natural than the same setup with well-designed second- or third-order crossovers. At the time, we didn't know why this was true. We were just getting started and first-order crossovers weren't something that had been written much about in audio reference books of the day. But a little research and we found what still seems to be the important thing today -- first-order crossovers are the only type which do not introduce time delays or phase distortion.

dB: *Why is time and phase distortion so noticeable once you make comparisons to sound produced with and without time and phase distortion present?*

RV: The audio-products industry is fixated on amplitude-related performance of loudspeakers and electronic components. They all look for amplitude-based performance parameters like frequency-response curves. Consumers have been lead down this "amplitude response is everything path" forever by the magazines and manufacturers. So it's hard to get people to understand that there are other factors that are even more important than amplitude response.

Early on, it became obvious to me from the testing that we were doing that the ear-brain mechanism could easily forgive minor, up to even modest narrow-band amplitude imperfections. But in the time/phase domain, even small amounts of time/phase distortion would greatly affect the live versus recorded test results. In the 20 to 30 years that have passed since our early experiments, I've become even more convinced that time and phase performance are far more important than the high-end industry has recognized. All of this knowledge we accumulated about the importance of very low levels of time/phase distortion came about as a result of those first experiments which were really looking for answers to other questions -- how to reduce and eliminate secondary distortions in loudspeakers.

dB: *What are we missing or what don't most people know or understand about the relationship between the audibility of amplitude differences versus time/phase differences?*

RV: Here's an example: If you set up a test so that you can vary frequency response over a controlled frequency band, say half an octave. And you set the test up very carefully so that when you increase or decrease the sound level in this half octave, that there is no change at all in the time and phase domains, you find that 3dB differences are just barely audible. But if you make a 0.5dB change and shift the phase at the same time, the difference is immediately noticeable. Only when phase or time delay were changed along with amplitude did the 0.5dB level changes become obvious. The digital test equipment that has been available for a while is really useful for this kind of testing. You can select any level change you want and get it with zero time/phase changes. You can also select time delays and phase shifts with or without level changes. It was harder to do this kind of testing in the pre-digital days, but we'd done some things just well enough to know we we're on the right road. Today's digital test equipment just helps prove the point even more convincingly. Time/phase distortion is simply more audible to us than most of us realize or recognize. The magazines and most manufacturers don't get this yet. [grins] After all, it has only been 20+ years since time delay and phase shift were looked at seriously by us and some other companies. [still grinning]

dB: *What can you say to people at home reading this who say, "3dB? Hell, I can hear MUCH smaller changes than 3dB. The stepped attenuator on my preamp has 0.5 (or 1 or 1.5dB) steps and I can hear each and every step completely and obviously."*

RV: Well there you're talking about a full-range change from 20Hz to 20kHz. And I agree with that, 3dB is easy to hear if it is full range. The ear-brain discrimination mechanism has more trouble when you limit the range of frequencies altered to half an octave or so. Our panel tests proved that this discrimination threshold varies from individual to individual a little bit. I believe that given enough time and familiarity, most of our panelists would eventually get quite a bit better at discrimination thresholds, but it's something you have to give people a means to do accurately then give them a lot of time to work with it in a system and room that they are very familiar with. Test conditions mask about half of people's perceptive abilities.

dB: *When doing the tests you were doing with amplitude and time/phase audibility, you varied the sound level 0.5dB over half an octave. Did you also introduce the time/phase shift only over the same half octave?*

RV: Yes. And going back to the issue of the audibility of a 3dB overall sound level shift, if you did that using a continuously variable pot you'd get a lot different results than with a stepped attenuator. Turning up a continuous pot very slowly is not immediately obvious and most people won't notice a level increase until you get to 3dB or so -- but only if you adjust the level slowly.

dB: *What makes loudspeakers with first-order crossover slopes and with no time or phase distortion harder to make than speakers which ignore time and phase distortion.*

RV: The disadvantage first-order crossover designs have is the large area of overlap between drivers. Unless the drivers remain in a completely pistonic mode through that entire large operating range you have to manipulate the phase and amplitude response within the crossover, within the driver and within the loudspeaker enclosure to get the drivers to produce the flat phase, time and amplitude response pattern you want at the listening position. For the drivers to remain completely pistonic over their entire operating range means there can be no cone breakup or resonances over that entire operating range. This isn't an easy thing to achieve, and even the best drivers will need some help from the crossover, enclosure and special attention to design and materials selection to be good enough to work well in loudspeakers with first-order crossovers. An interesting side effect of driver development for first-order crossovers is that as you eliminate more and more of the shortcomings of the driver in the problem areas, the performance of the driver improves noticeably in non-problem areas too.

You can't just take two or three or four drivers and put first-order crossovers on them and have a time and phase correct loudspeaker. In a two-way speaker, first-order crossovers mean a single capacitor in series with the tweeter to roll off the bottom end of the tweeter and a single inductor in series with the woofer to roll off the top end response of the woofer. If you put normal drivers (off-the-shelf) in this two-way speaker, you'd more than likely have to wire the woofer and tweeter with opposite electrical polarity to get anything approaching linear amplitude (frequency) response. But it's my opinion that putting drivers out of polarity in a loudspeaker is something you do in a cheap speaker, not in high-end speakers. Imagine a crossover at 3kHz and a sound being reproduced at that frequency. Half the energy (in our two-way example) will come from the woofer, the other half of the energy will come from the tweeter. One driver will be moving forward while the other is moving backward. You get a full or partial cancellation of that sound because of the polarity reversal. As you move away from the crossover point, the amount of cancellation decreases. To get those two drivers working in the same polarity requires a crossover with a lot of additional components and probably some significant mechanical or material design changes to the driver itself. But that's what high-end design is all about -- doing the design the right way to end up with a product with the best possible performance in every possible way.

If you are using second- or third-order crossovers and one of the better computer programs for speaker design, you can plug in stock drivers with the right specifications and assemble a crossover that that works well with the drivers and end up with something pretty close to a finished product. First-order crossover designs aren't that easy. It takes a lot more work and *incredible* demands on the drivers themselves. It isn't that doing the work on a good first-order speaker design is impossible or anything, but it is expensive and takes a lot more time than designing loudspeakers with second-, third-, or fourth-order crossovers. I look at this as another thing that separates high-end designs from other products: True high-end designs, to be worthy of being called that, should be paying attention to the difficult details.

dB: *There are some loudspeakers being made which have physically staggered drivers, the old "line up the voice coils" look to them. They either slope the front baffle or make a series of progressively smaller sub-enclosures which are staggered in space. But they are not time and phase correct loudspeakers because they do not employ first-order crossovers. Just how far would drivers have to be staggered to eliminate the time delays in second-, third- or fourth-order crossovers?*

RV: Well it certainly isn't practical, but I think I understand the point of your question. If the crossover was fourth order, the distance between drivers would be huge. Some manufacturers confuse the "time and phase correct" phrase for the consumer, making it harder to understand what their speakers are really doing. They will claim that their speakers are time and phase correct if they get all the drivers working in the same electrical polarity. Most of the speakers made today have every other driver connected with reversed electrical polarity to get the frequency response to look reasonable. With that kind of speaker (mixture of driver polarities) you could never get the speaker into absolute phase because you could only get every other driver into absolute phase at any one time. So in the last ten years or so, some loudspeaker designers have confused the issue by building, for example, a fourth-order design where they have worked hard to get two or three or four drivers connected with the same electrical polarity. They then refer to their design as time and phase correct.

To my way of thinking this is really misleading to the consumer since these fourth-order designs still contain large amounts of phase shift and time delays between drivers in spite of the fact that all the drivers are in the same electrical polarity. To have a truly time and phase correct loudspeaker design, you should be able to input a broadband square wave into the loudspeaker and get a triangular wave out of the speaker (because loudspeakers can't do DC to light, so the edges of the square wave will be rounded or angled off by the speaker). The time and phase correct fourth-order loudspeaker will not reproduce this broadband square wave pulse with anything approaching the coherency of a true time and phase correct loudspeaker with first-order crossovers.

If you look at the impulse response graph in *Stereophile* magazine, this is exactly the kind of response I'm talking about. The leading edge is very steep, just like a square wave with the tweeter response starting right at the top of this vertical line. In a first-order loudspeaker with correct time and phase response, the midrange driver's response connects to the tweeter's response and gradually decreases in level on an angle (a slope). The woofer's contribution is next and it further continues the slope of the falloff. If you send the same impulse to any speaker which does not have correct time and phase response, you'll see one of two things: separate responses for each driver but all in the same polarity or a positive tweeter response followed by a negative midrange response because the phase response is radically different, 360 degrees or more phase shift, followed by a positive woofer response. The latter examples are not reproducing the impulse accurately, so those loudspeakers cannot reproduce music accurately. The flatter the sloped line is in the impulse test graph, the flatter the frequency response of the loudspeaker being tested. A lot of people don't understand the importance of the impulse test and the resulting graph of impulse response as it relates to audible performance of the loudspeaker.

dB: *Looking at various Vandersteen loudspeakers, you can see that there isn't more than 2" to 4" of stagger in the positioning of the drivers to achieve the correct time-aligned position of the drivers. Explain two things: What are you really lining up by staggering the drivers and how big the stagger between drivers would have to be in speakers with something other than first-order crossovers.*

RV: Well, the stagger distance would be very large for second-, third-, or fourth-order loudspeakers. With narrow and deep loudspeakers being fashionable these days, there *might* be enough depth in some of them to physically stagger the drivers enough without making the speaker any deeper. But that much stagger introduces such severe diffraction problems that it's really impossible to make a commercial product that way. As far as what we are lining up, it isn't the voice coils as consumers have been taught by the magazines. We're really lining up the acoustic center of the drivers. This is an imaginary point where the sound can essentially be said to emanate or originate from. This point is different for different drivers, so you have to figure out where this acoustic center is for each driver in each different loudspeaker model you use it in. The design of the crossover is a significant element of determining just where that acoustic center is located. So you cannot just make a blanket statement that you "line up the voice coils" to get the drivers staggered correctly. If fact, it's unlikely that lining up the voice coils would give you the right amount of driver stagger.

Our stagger is part of our minimum baffle design. The baffle is the flat surface the driver is mounted on. We try to make that surface as small as possible because our research proved and keeps proving as we revisit those tests that the smaller the baffle, the less surface there is around the driver, the better the driver sounds. Using foam or felt around the driver to attempt to reduce the effect of the baffle is better than nothing, but no baffle at all is much better. This is a direct result of learning about diffraction and how detrimental it is to the sound of loudspeakers by introducing large amounts of the secondary distortions we set out to eliminate in our speakers way back in the '70s. Small or no baffle produces no diffraction distortions. We put each driver in a separate enclosure so we can minimize the size of the baffle around the driver.

Internal diffractions or reflections are another thing we try to control. We use long damped transmission lines in an attempt to reduce the amount of energy that can bounce back against the rear surface of the driver and cause it to move in the absence of an electrical signal. That reflected energy, whether from a baffle or internal reflection, is not a part of the original input signal, so loudspeaker designers should be doing all that they can to keep those diffractions and reflections from contributing to the sound heard at the listening position.

Those sounds are in fact 100% distortion. They weren't in the original input signal; they originated at the loudspeaker as artifacts of a large baffle or poorly controlled internal reflections. Another trick we employ and have patented is reducing the diameter of the magnet structure and shaping the magnet structure of the driver. In conventional drivers, pressure waves coming off the back of the driver's cone do hit the magnet and reflect right back onto the driver cone and cause the cone to make a sound that should never be there, another secondary distortion. We have been eliminating more and more of this reflection as we refine our drivers over time.

dB: [laughing] *Richard, this could be something that will go right over people's heads. I mean, it's obvious to me what you are saying is true, but you said it in such a simple way that the impact, the importance of the facts, isn't going to be fully appreciated by a lot of people reading this interview. I hope they do get it though. It's a significant fact of life that is not being addressed very well out there in speaker-land.*

There are people, smart people, in various aspects of high-end audio from publishing to manufacturing to research who are convinced that time differences smaller than 4 or 5 milliseconds (.004 - .005 seconds) are inaudible. Most of the time delay artifacts from loudspeaker crossovers and lack of stagger of drivers fall within this 4 to 5 millisecond window so people think their designs are as good as they can get as long as they are within the window. Let me do the math quickly -- using 750 mph as a rough equivalent for the speed of sound translates to approximately 13" of physical distance for each millisecond of time delay. Five milliseconds would translate to just about 5 1/2'. To me it seems incredible that the time delay between two sources separated by 1' to 5' to my ear would be inaudible.

RV: There are two things going on here. Number one is that there are people who can't hear differences between amplifiers or interconnects or speakers cables or different CD players doing the research that leads to bogus rules of thumb like this 4 - 5ms window being the threshold of audibility. That doesn't mean an audible difference doesn't exist; it just means the people doing the research honestly did the best they knew how and ended up with a result that isn't accurate. How could you expect a different outcome? They certainly would never have made the startling discovery that 1ms or 0.5ms is the real threshold of audibility.

Number two has to do with what goes on with many of these tests that some sharp people have accepted as being true. These are, after all, tests. When you put human beings under test and they know they are under test, results change. Human beings under test do not react to those things our right brain tells us are important under relaxed, non-test conditions. We respond to things very differently when we are in our own listening rooms listening to our own equipment and our own favorite music. In that environment, our right brain becomes an integral part of the listening experience, and our discrimination faculties are doubled or squared.

It's very difficult to take a person who is known to be a good and experienced listener and put him in an environment where he knows he is being tested and get anything remotely resembling the results he gets in his own familiar listening environment.

Fundamentally, it comes down to the fact that the tests that give us all of these magic numbers are flawed. None of them have done anything to get the test subjects into the proper state of mind. By that I mean, as conducted, the tests exercised only the analytical side of the human mind. The thing that we need to make the test accurate is unavailable to us in the test environment because of the way human minds work. Put us under test and the right brain takes a hike and leaves all the work to the analytical left brain. So far, in these test environments it has been impossible to get people into the same relaxed state of mind that they are in when they are out enjoying live music or when they are listening to their own systems for pleasure.

Many audiophiles suffer from this too. They can't stop analyzing their system and enjoy music. They have to constantly compare new components or wires. They tweak incessantly. They clean connections. They are constantly fiddling. These people can't ever let their right brain loose long enough to just sit down and enjoy the music. Their left brain is in overdrive all the time.

dB: *Perhaps this is why a certain herb is so popular among some audiophiles* [grins] -- *it beats the left brain into submission and lets the right brain come out to play. Engineers, scientists, and other technical types seem to benefit the most while artists, musicians, and fiction writers wonder what the big deal is.*

RV: OK. But I can't endorse that kind of thing though, you know. [grins] If you ever find yourself in "test conditions" and unable to hear differences you know you should be able to hear, try this:

Stop trying to quantitatively analyze the bass, the midrange, the dynamics, the transparency and all that and concentrate on your emotional response to the music being played. Remember your feelings about the music and you'll find that all of a sudden, differences exist where they did not exist a few minutes ago. If you keep trying to be objective about the sound quality, everything just runs together and you find yourself unable to form an opinion.

- **Continue to Part Two**

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Richard Vandersteen of Vandersteen Audio: Part Two
by Doug Blackburn (db@soundstage.com)
Continued from Part One

dB: What's different about time and phase performance when you use a concentric driver (i.e. tweeter mounted in the center of the woofer cone) like you did with your center-channel and surround loudspeakers?

RV: Concentric drivers are interesting -- you can make them time/phase correct everywhere except dead center on the axis line of the driver. So you don't want our center-channel or surround speakers pointing directly at the listener's ear(s). You want to be a little bit off-axis. This actually works extremely well for center-channel and surround speakers. With a center-channel, the loudspeaker is above or below the image which puts the center speaker above or below the viewer's ears. With surround speakers, they are usually above and/or behind the listener. We think a full array of time/phase correct loudspeakers makes the best possible home-theater setup. Something happens that you just can't get with typical loudspeakers. In fact, we are getting quite a bit of notice lately from the home-theater crowd because or sound is so much different than typical home-theater setups.

dB: How much of a change would you hear if you get outside the time/phase correct window for one of your loudspeakers?

RV: Let's use a three-way as an example. At most listening positions, there's a window about one foot high where you are aligned with the loudspeaker. If you are a little low, the top end will sound rolled off and the bass will be a little prominent but the sound will still be pretty linear, if tilted a bit. The worst case is standing so that you get the sound from the tweeter first. In that case you'd hear a 6dB to 8dB trough in the midrange and a 5dB to 6dB peak in the tweeter. That would be a very audible change. It's also one of the reasons I hate to see people listening to our loudspeakers while they are standing up. You just can't make any accurate observations about our sound until you are seated in that time/phase correct window. But this happens all the time at shows and at dealers. It's something we have to live with to make the products we make.

dB: Speaking of auditioning loudspeakers, what can people hear or listen for in a time and phase correct loudspeaker that they won't hear from typical loudspeakers?

RV: Well that's a bit of a problem because there are only four or five brands of loudspeakers being made in the world that are truly time and phase correct. One thing you definitely want to be aware of is, if you are listening to a new loudspeaker and it just literally blows your socks off, it just really gets your attention and you come out of that demo saying something like "Wow, I've never heard anything like that before!" That should be your first indication that something is wrong. Loudspeaker sound/performance does not change that rapidly, at least not as things are in 1998. For some loudspeaker to be that different to an experienced listener compared to anything else they've ever heard -- somebody's playing games. There are undoubtedly going to be some very attractive peaks and dips in that loudspeaker's response in order to get it to be that noticeably different. A really good time and phase aligned loudspeaker will leave you very unimpressed with the loudspeaker, but very involved with it, very drawn into the music. The sound should sound less hi-fi (perhaps less audiophile) and more like live music. Even recordings that are not all that well done technically should sound at least entertaining and listenable. You should find yourself transported to the performance or find the performance transported into the listening room depending on the perspective of the recording. That's the best way to "get" time and phase correct loudspeakers. Listen for the emotional connection with the music and stop trying to enumerate the Top 10 Audiophile Sound Characteristics (i.e. transparency, dynamics, bass, mids, highs, tonality, harmonics, etc.). Not that those things are unimportant, but if you never stop listening in that way, you'll never notice the other side of loudspeaker performance. A realm of performance that is often completely ignored by dealers, reviewers and audiophiles.

The soundstaging of time and phase correct loudspeakers tends to be more natural also. You'll get randomly spaced images of performers spread from wall to wall. Other loudspeakers tend to bunch the sound so that everything happens between the loudspeakers. This is often described as the speaker having a strong sense of focus. All this means is that the loudspeaker in question doesn't have the ability to produce a panorama of sound across the room like a good time/phase correct loudspeaker. To one degree or another, the sound favors the center and doesn't have the expansiveness that good time/phase correct loudspeakers have.

Q-Sound[TM] is another area where time/phase correct loudspeakers will do things that typical loudspeakers can't do. Most typical loudspeakers will keep the time and phase manipulated Q-Sound effects in the front half of the listening room. Time/phase correct loudspeakers will put some of the effects in their correctly intended locations behind the listener and well off to the sides of the room behind the listener making a 360-degree sound field. Typical loudspeakers may crudely position some of these sounds behind the listener's head, but they will not have the lateral specificity that they have when played back on time/phase correct loudspeakers.

dB: Why are all Vandersteen loudspeakers made with a single driver covering each frequency range?

RV: From 150Hz on up, single drivers are the only way I know to make a true time and phase correct loudspeaker. Below 150Hz, the wavelengths are long and the events in the sound are long, so using multiple drivers down that low is not a problem. From around 150Hz and higher you must have only one sound source. This is because it is impossible to get two exactly identical drivers to cover a frequency range and because they will be physically separated in a vertical array, it is impossible to get the distance from each of those drivers to the listener's ear to be exactly the same. Because of those differences in driver response (even if matched to small levels) and different driver-ear distances, time smear is introduced. If you're shopping for a time and phase correct loudspeaker, you don't want to be paying big bucks for a design that undoes much of what is good about time and phase correct loudspeakers.

dB: Can panel loudspeakers be time and phase aligned?

RV: For the most part no. Panels tend to be large and the distances from the panel to the listener's ear are different for each point you measure. A sound radiating from a panel has many points of origin on the panel and those all have different distances to travel to get to the listener's ear. So panels introduce quite a bit of time smear for the most part. An exception is the Quad 63, which mimics a point source because all the sound radiates from the center of the panel. You can physically observe phase problems in panel speakers by shining a light on the panel and observing the reflection on it. The panel will shimmer. It's obvious after seeing this shimmer that the panel is not operating as a true piston. It is a light membrane, but the field acting on the membrane is very weak compared to the field in the voice coil of a good dynamic driver. Because the magnetic field is weak, there isn't a strong degree of control of the membrane. That shimmering that you can observe is what gives panel speakers the sound that some people like. It's like having thousands of tiny low-level reverbs added to the music. They give a sense of space like reverb on recordings does. The best thing about panel speakers is that they have nothing that reflects the back wave back onto the membrane. The backs are open so the back wave can get away from the speaker without causing any secondary distortion. We try to mimic that with the enclosures we design for our drivers and by constructing our drivers so there are no surfaces behind the cone for sound to reflect off of and back onto the cone. This approach eliminates a lot of time smearing and amplitude problems.

We've been working on reducing the back-wave reflection and reflections off of driver parts for 20 years and have some patents on our methods. Now you are seeing some major players incorporating these features in some of their very expensive loudspeakers. I guess that means that more than one little loudspeaker company thinks it's an issue.

dB: Why is it so hard to integrate a dynamic woofer with a panel loudspeaker design? I don't think I've heard one yet that sounds right.

RV: It is a very difficult thing. For many years, there seemed to be some kind of "ideal" loudspeaker which had the positive characteristics of a panel speaker with the positive characteristics of the best dynamic woofers. The difficulty is that if the crossover is done anywhere near the midrange frequencies, say anything over 100Hz, your ears are very sensitive and regardless of how good a job you're doing with your panel and your woofer separately, your ears will always pick-up on the fact that the sounds are being reproduced differently. Our ears are often not real accurate when we look at a specific point. But our ears do a heck of a job when they have to relate one sizeable chunk of the frequency spectrum to another large chunk. Maybe I'm overstating this a little. It's something that we tune into over time. We might not notice it in a 15-minute demo or even in a whole day with such a loudspeaker. But given days or weeks, it is something our ears will pick out. I believe you can integrate a woofer with a panel, but only if you do it at very low frequencies -- 80 Hz or lower would be a good crossover point for that kind of design. But this means a huge panel with flat response to 40Hz in order to be able to mesh with the dynamic driver. When building to a price point, panel size is usually one of the first things to be reduced and as you do that, the crossover point has to go higher and higher till you get into the trouble zone at 100Hz and above.

dB: What are some of the things about the drivers used in Vandersteen loudspeakers that make them suitable for use in time and phase correct loudspeaker designs?

RV: Our drivers are our single greatest technical achievement. Nothing we can buy off the shelf can come close to what our drivers do for us. They literally make it possible to design and build loudspeakers with first-order crossovers without severe compromises. Dealing with the rear-wave energy is probably the most significant single technical feature of our drivers. There is just as much energy coming off the rear of the cone as comes off the front. What you do with that energy is critical. When you see a typical midrange driver, you see a structure that looks a lot like a cup with a few holes in it. All the energy reflected by the basket and magnet are reflected right back onto the cone giving a source of 100% distortion -- a lot of time smear and amplitude problems. This is a pretty severe distortion. It tends to make midrange notes harsher than they should be. Typical loudspeakers and drivers only attenuate this rear wave reflection 3dB to 6dB. If your midrange driver is cranking along at 85dB, most loudspeakers are hitting the back of the cone with 79dB to 82dB of reflected sound from the driver and/or enclosure. This is very significant and gets to be easy to hear when you are familiar with the sound of loudspeakers that do not have as much internal reflection. In the live versus recorded test it is especially and immediately obvious.

The average audiophile listening to loudspeakers is at somewhat of a disadvantage because they don't have the original sound in the room for comparison. People tend to get used to the way loudspeakers sound and most loudspeakers have very similar levels of distortion (for like types of loudspeaker). This gives audiophiles the impression that a certain kind of sound is "correct" because the reference point shifted from live music to comparing one loudspeaker to another. When you hear a loudspeaker without all that rear wave distortion, it will sound like the notes are coming out of a blacker background. There's less hash -- it's less harsh. Some people will even perceive the lower distortion loudspeaker to be less transparent or a little sweeter sounding than it ought to be. If they can get past that initial opinion which came from comparisons with other loudspeakers with higher levels of distortion, and live with one of our loudspeakers for a while, they will find that the sound is more like what they experience in a live setting and less like the sound of a loudspeaker.

dB: Where did the transmission-line enclosures and separate enclosures for each driver come from?

RV: We use a terminated transmission line behind the tweeters, midranges and woofers. These aren't completely open like a true transmission line. What we try to do it direct that rear-radiated energy away from the driver in as long a path as we can. When the sound reaches the end of the long chamber, which is not in a straight line, it has to pass back though the maze losing energy at each reflection point and through the damping material. We manage to attenuate the rear wave energy to the point that it is inaudible. That's probably one of the hallmarks of our speaker designs. Separate enclosures, I think, is pretty obvious. You can optimize the enclosure for one specific driver and you can maximize the absorption of the back wave when you are dealing with a known frequency range within the separate enclosures.

dB: You have an interesting perspective on a long-time audiophile tweak -- removing the stock internal speaker wiring and replacing it with high performance audiophile cable/wire.

RV: A properly designed crossover is an all-encompassing kind of thing. Every detail matters when designing the crossover. This means that the enclosure and driver itself affect the crossover design. The interaction between drivers affects the crossover. The length and gauge and other electrical properties of the wire connecting the drivers to the crossover also affects the design of the crossover (if the designer is doing a thorough and detailed crossover design job -- a lot of them do not). To get all of these elements to do what you want to have happen at the listening position is an incredibly complex and time consuming task. Change the wire and you literally change the loudspeaker into something else very different -- at least this is true for Vandersteen loudspeakers. So an audiophile could go into a Vandersteen loudspeaker and install what we would all agree is a superior wire and end up with a loudspeaker that sounds worse. Not because the replacement wire was bad, but because we'd have to rework the crossover to incorporate that specific wire to overcome the other problems that were induced by changing the original wiring. And this wouldn't be the kind of crossover reworking you could do with a computer program. This is the kind where you spend tens or hundreds of hours testing and experimenting to get the fine tuning of the crossover correct.

This wiring issue is one reason no Vandersteen loudspeaker has ever had one inch of "full-range wire" in it. "Full range wire" means wire that carries the audio signal from the amplifier -- the entire audio signal from lowest bass to highest highs. We avoid this by putting our loudspeaker terminals right on the back of the crossover. When you tighten down the terminals or insert the banana plugs, you are plugging your speaker cable directly into the crossover. Our speakers all have the terminals up about mid-height so that the wires to each driver can be of similar lengths and therefore, similar electrical characteristics. All of the wire in every Vandersteen loudspeaker is after the crossover to connect the driver to the crossover. This (enclosure, driver, wire, and crossover) becomes a network that has to be tuned. The crossover is where you do the tuning.

dB: Over the years there has been a list of audiophile complaints about Vandersteen loudspeakers. [slight chuckle] Let's talk about some of these and put them in perspective.

RV: [more obvious chuckle] OKaaaayyyyyyyy.

dB: Let's talk about appearance first. A Vandersteen loudspeaker made in 1998 looks an awful lot like a Vandersteen loudspeaker made in 1980. People like things that look new and different. There must be some pretty good reasons why the speakers look the way they look and why the appearance doesn't change much over the years.

RV: The structure of our loudspeakers is an important part of why the speakers sound the way they do. The average loudspeaker sold today has 70% of the parts and labor cost in the cabinet and finish. Unfortunately, the prettier and shinier that cabinet is, the more it is going to interfere with the sound quality. You can't get away from this -- it is pure and simple physics. Our cabinet is very functional. In effect, our cabinet is designed not to be there as far as the sound of the loudspeaker is concerned. We do everything we can to make the cabinet acoustically disappear. That's what our boxless design is all about. Our cabinet and finishing represents about 17% of the parts and labor cost for our loudspeakers. The typical loudspeaker has 30% of the parts and labor cost in drivers and crossover. Vandersteen loudspeakers have 83% of the parts and labor cost in the drivers and crossovers. This is why we have achieved a reputation for building products that are incredible values for the money spent, even among critical audiophiles. Our enclosure isn't very pretty to look at without the grille cloth in place. But that construction method allows us to create amazingly high-tech cabinets without having to worry about making them pretty which increases the cost of the loudspeaker beyond what I think is reasonable.

There is a growing number of narrow loudspeakers. Some people wish we'd do a narrow design. We just can't figure out how to do that without doubling the cost of the speaker or compromising the sound quality and I won't do either of those. One of my goals for Vandersteen products is to give people the most performance I can for the money. I'm just very uncomfortable putting too much money into the enclosure. The Model 5s are an exception. From 1977 when we started we have always promised Vandersteen dealers and customers that we would not spend one penny on the looks of our loudspeakers if it didn't enhance the performance of the loudspeakers. The Model 5 is different because it is so expensive to manufacture that it sells for a price high enough to leave some room to do some things appearance-wise. So when we did the design of the 5s, we put in everything we knew to do in the design of a loudspeaker. We optimized its performance in every way that we could think of. Once we had the technology package, then we put some money into the appearance of the loudspeaker. But even with the Model 5, appearance is the lowest-priority element of the product.

dB: Another perennial comment about Vandersteen loudspeakers you hear repeated over and over again from certain quarters is that they sound dark and slow.

RV: [chuckles at the "dark and slow" phrase] Let's touch on the "dark" aspect first. Our top-end response is flat all the way out to 20kHz or 30kHz depending on the speaker model. We have single drivers for each frequency range. They are time and phase correct. We work very hard to make the bass and midrange response as flat as possible. We do not put intentional peaks in our tweeter response. Yeah, I understand why that [dark and slow] comment gets made a lot. In a dealer showroom, there are going to be other speakers with tweeters playing 3dB to 6dB louder than ours. So, when you put us up against those kinds of loudspeakers, yes, we sound a little dark, a little laid back. But when you compare us to real music, we're right there. Those peaked tweeters are completely intentional and are designed to cause the customer to spend money on the speaker with the peaked tweeter rather than on the more natural sounding loudspeaker. I could fight back and put 3dB to 5dB more into our tweeters, but I think it is the wrong thing to do and I'm going to keep building loudspeakers as honestly as I can. Our loudspeaker with flat response does not sound very flamboyant next to an intentionally manipulated loudspeaker.

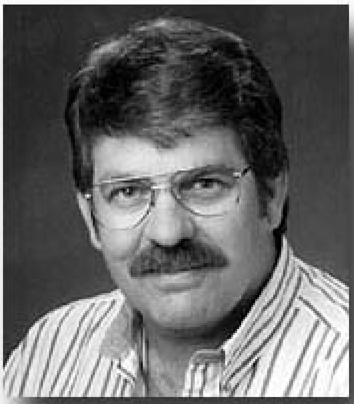
There are all kinds of other potential problems in dealer environments -- other speakers in the room, inappropriate cables (perhaps better suited to other loudspeakers with peaked tweeters), inappropriate electronics, poor setup, incorrect connections. We try to train our dealers well, but things happen. Even more to the point, the dealer's show room is often not an ideal listening environment due to the realities of it being a show room and not a listening room. Most of our dealers do a heck of a good job with their demos given what they have to work with which is more limiting than a home listening room. It isn't that hard to hear a Vandersteen demo where our loudspeakers sound a bit boring next to other loudspeakers.

But if you optimize a system for Vandersteen loudspeakers and put them in a room by themselves you get a listening experience that is very different. People who know Vandersteens have always felt that way and have disagreed with the dark/slow comment that keeps being repeated. People who make the dark/slow comment, I think, have never heard a pair properly set up in the home environment where they were designed to function correctly. We never designed a loudspeaker to sound good at a dealer show room. I don't want loudspeakers staying in dealer show rooms, I want them in peoples' listening rooms -- and I want them to get there without playing tricks on people.

What you don't hear much comment about from the same group that keeps repeating that dark/slow criticism is the number of $1200 Vandersteens out there being driven by $5000 to $10,000 worth of electronics. This isn't accidental. It would never work if the loudspeaker wasn't giving performance well beyond its price. This is something the dark/slow skeptics can't understand. This is a direct result of our strategy for our products.

- Continue to Part Three

December 1998
Richard Vandersteen of Vandersteen Audio: Part Three
by Doug Blackburn (db@soundstage.com)
Continued from Part Two



dB: Let's change the subject a bit. What's different about the Model 5 loudspeaker compared to other $10,000 loudspeakers that don't have power cords coming out the back of them?

RV: [laughs] First, the Model 5's starting point was everything we learned doing all of our other loudspeakers. If you take the five-sided grille cloth cover off the head of the Model 5 and ignore the beautiful wood finishes we use, you'll find that the working structure of the 5 is an evolution of every Vandersteen loudspeaker ever made. The head of the 5 is a Star Wars-looking thing with a tweeter, a midrange and a woofer. Each driver has a separate transmission line that is better at damping the back wave off of the drivers than any loudspeaker enclosure we've been able to create before. The tweeter and midrange driver are in terminated transmission lines, which means they are sealed, labyrinth-like passages that contain damping materials. The shape of the transmission line is very irregular. This combines with the damping materials to reduce the energy in the back wave from the drivers. The woofer is in what we call a resistive transmission line, which I don't want to go into too much detail about. It's a very special enclosure that lets us do things with the woofer that we could not do otherwise. We do as many things as we can to eliminate the baffle, the flat areas around the drivers, to minimize the size of the baffle in relation to the size of the driver. Like all the other Vandersteen loudspeakers, the crossover uses only time-and-phase-correct filter networks.

The real essence of the Model 5 is that it is a cost-no-object Model 3A with a cost-no-object 2W powered subwoofer, all built into a single very elegant package. The cost-no-object factor let us do some more extravagant things to make the enclosure as inert as possible. We were able to play with dimensions and get a width and depth that make the speaker very stable in all directions. Of course we were able to do things with the drivers that go beyond even what we are able to do at the price points of the 3A and 2W subwoofer. We believe that the Model 5 contains the most advanced drivers in the world.

We were able to enhance the power and performance of the amplifier that powers the self-contained subwoofer. We retained the connection for the subwoofer to the main amp that we use with the 2W. This way, no matter what main amplifier you use, the bass of the Model 5 will maintain the same character as the bass of the main amplifier making the subwoofer section more coherent with the rest of the loudspeaker. We were able to spend some money on tooling to build some of the proprietary pieces for the Model 5 that help to

make it something really special.
We have what we like to think of as a buyer-protection plan. The Model 3A and 2W for example, we promise customers that they will be able to upgrade both of those products in the future as improvements are made. Because of the way the loudspeakers are made, this usually means the speakers have to be sent back to the factory for the updating. So it isn’t something people will do often, but it is there and we do this kind of work regularly. Many pairs of Model 3 loudspeakers have been upgraded to Model 3A, which is a significant upgrade, for example.
The Model 5 continues this upgrade/update guarantee path, making it a solid long-term investment, but we made it even better. We guarantee Model 5 customers that all upgrades and updates, whether to drivers, crossovers or amplifier are able to be performed in the customer’s home with a Phillips screwdriver and a soldering pencil. We realize this is a very expensive purchase for many of our customers and we believe that protecting the customer’s purchase with an in-home update/upgrade program is something our Model 5 customers deserve. Most audiophiles who have been doing the high-end thing for any length of time have been through the pain of not being able to get more than half of their original purchase price for a component that is only a year or two old. Doing that very often with products costing $10,000 or more would tire me out, so I did as much as I could in the Model 5 to avoid putting our customers through that.
All the things the Model 5 does so well have been dealt with in our other loudspeakers. It’s just that in the Model 5 it’s all taken to the extreme.
dB: I’ve heard comments from people a few times about a couple of things they don’t like about the Model 5. One criticism was having the 9-volt batteries that bias the capacitors in the crossover soldered into the circuit so that owners can’t access the batteries and change them themselves. The other thing was the setup of the bass response of the subwoofer, which apparently requires some special training to do correctly. The concern being that if owners want or need to move the Model 5s after the dealer has installed them, they wouldn’t be able to get the bass right without the dealer coming back to adjust the many poles that help contour the response of the subwoofer.
RV: We have been able to refine the setup of the bass contour controls significantly since the introduction of the Model 5s. The owner’s manual now contains what we think are very clear instructions on how an owner can do these adjustments himself. It does require the use of a real-time analyzer, which is easy to rent from many sources in many cities -- even by mail. But our belief is that the customer who spends that kind of money on a loudspeaker deserves to be able to call the dealer and have him come by within a reasonable time window and do the setup. It isn’t like the bass is going to completely fall apart if you move the speakers six inches or a foot, so you don’t necessarily need 911 bass paramedics to show up every time you change the speaker position a little bit. We couldn’t get the subwoofer to integrate properly without doing the bass contouring the way we did it. So basically there would have been no Model 5 without the bass contouring the way it is. Other manufacturers chose to make essentially non-adjustable bass response in their flagship products. We chose something more complicated that might be a small inconvenience a couple of times over the life of the loudspeaker, but the results are unlike anything else on the market.
We felt that having the dealer do the bass setup on the Model 5 was the best route to take because it does take some time. You get better at it with practice too. Since dealers see many more pairs of Model 5s than the owner of a pair, the dealer is the person who is going to get all the setup experience. But our instructions in the owner’s manual do work and we do have customers doing their own bass alignment using a rented or purchased real-time analyzer.
The crossover bias batteries have an expected lifetime of at least five years. There is no current draw on the batteries. So the lifetime is essentially the same as if the batteries were sitting on a shelf in their unopened package. The batteries must be soldered in because any noise at all at that connection produces a 9-volt transient. If you put 9 volts into any modern amplifier, the amplifier thinks you are trying to reproduce an earthquake and it will do its best to bring down the house. This will cause serious, serious damage to the loudspeaker. Most amplifiers are driven to full power with 1.5 volts or so. So you can only imagine what 9

volts would do to the subwoofer amp. I have no problem with knowledgeable customers soldering in replacement batteries into their Model 5s as long as they understand what they are doing. They need to have soldering skills and realize that they have to unplug the Model 5s from the wall so that the subwoofer amplifier is off when the batteries are changed. Model 5 owners without the technical skills to replace the batteries should ask their dealer. Again, nothing catastrophic happens if the batteries die. The Model 5s keep on working fine. There is a very small degradation in sound quality, but certainly nothing serious. The bias batteries are one of those cost-no-object things that enhance performance a little bit.
Nobody has complained to me about either one of those features of the Model 5, by the way. [laughs]
dB: Must be some picky audiophiles looking for excuses. [both laugh]
dB: Today you see a fair number of loudspeakers being made with minimalist crossovers, perhaps a single capacitor to roll off the bottom end of a tweeter and a single inductor to roll off the top end of a woofer. Or in some cases, no crossover components at all, relying on the characteristics of the driver and enclosure to roll off the bottom and/or top end of the driver's response. What's going on with these kinds of designs?
RV: If they use good basic drivers, it would certainly be better-sounding than a poorly done complex high-order crossover. That's a fact. One of the reasons these kinds of designs are so popular is because they remove tens or hundreds of opportunities for the engineer to make severe mistakes. Because of that, these minimal designs will tend, as a group, to sound pretty good. On the other hand, that minimal approach will never be a substitute for a properly designed complex crossover with good parts quality combined with excellent drivers. When you take the time to design the complex crossover well you can do more things. You can control things better. Nowadays with good film capacitors and air-core inductors it is possible to make crossovers for 150Hz and higher that have no insertion loss. You have to know how to engineer crossover in a different way though. You have to be able to design the crossover, treating the acoustic properties of the driver and the electrical characteristics of the crossover as elements of a single entity. Even the wire connecting the driver to the crossover is part of this entity and has to be accounted for in the design. You can actually surpass what you can do with biamplifying and an outboard electronic crossover.
[dB notes: This is the first time I've heard anyone who actually knows what is possible in both passive and active crossover realms make a claim like this. No longer are passive crossovers, 150Hz and up, necessary evils as we have been led to believe from our earliest audiophile days.]
Below 150Hz you have some problems because the magnitude of the impedance swings is large. You're looking at very large inductors, which get very cumbersome. You're also looking at probably having to use electrolytic capacitors because the capacitance values you need are so high. We overcome that by building the crossover response we need right into the amplifier that powers the subwoofer in the Model 5 and 2W and using simple capacitor as high pass filters for the bottom-end roll-off of the woofer. These low-frequency passive crossovers are still very difficult to do even with the best parts that are available today.
But above 150Hz, today's parts let you design a passive crossover that is far, far better than active crossovers if you know how to design the crossover to account for the acoustic properties of the driver and the electrical properties of the crossover as a single entity.
dB: Do you have any favorite music, CD or LP, you use for evaluating loudspeakers or systems that people could go out and buy today?
RV: I don't have anything special that comes to mind. But I have found that what gets you closer to the truth is getting familiar with a broad spectrum of music.
We use our own master tapes, not because they are so great or so well done, but because I know exactly what was going on when they were made -- that makes them the ultimate reference. I can't imagine being a speaker designer or manufacturer and having to rely on commercial recordings to reveal whether design A or design B in the speaker is more right or more wrong. You don't really know what the commercial recording sounds like -- you have no reference to it. You weren't there when it was recorded and mastered. We did turn two of our master tapes into CDs and LPs. They aren't as revealing as the master tapes, but they are

pretty good. Chico Freeman Saudades [Water Lily WLA-CS-16-CD] and Terry Garrison Only Love [Vandersteen Audio VA-CD1]. Their biggest merits to me are that I was there when they were made. I setup the recording system myself and sent clicks though it to verify that I was polarity correct through the entire recording chain. I also used a single stereo microphone to do the recordings to insure that there were no time and phase errors.

dB: Let's switch to home theater for a bit. What problems are people going to run into when they are trying to get a home theater that sounds good?

RV: A significant difficulty is getting people into five speakers and five amplification channels. Most people try not to increase their budget over what they'd be willing to spend for a two-channel system, so they usually end up with something pretty mediocre. People's financial situations haven't changed since we went from two channels to five channels, so their budget hasn't changed. My recommendation for them is that they build on the two-channel system they already have and not start all over again unless there are real problems or limitations in their two-channel system.

There are some people in the two-channel and home-theater worlds who believe that home theater is somehow completely different from two-channel music. To the point that they think good home-theater speakers have to be different than good music speakers. Sometimes they even think the electronics for music and home theater need to be different. I don't understand that thinking and I'm very much opposed to that approach. Movies are full of synthesized sounds, explosions -- who knows what they are really supposed to sound like. There's also a lot of dialog, a lot of music, people clinking glasses toasting one another, sounds that we're all very familiar with. We take the principals we've applied to the music world, especially the time-and-phase-correct nature of our loudspeakers, and extrapolate that out to five channels. We can do that movie more realistically than what the industry is trying to put on people with this mistaken notion that you need something different for home theater.

Home theater does need good subwoofers. Even our full-range loudspeakers cannot handle some of the cone excursions that result from some of the movie soundtracks. So even when you have really good full-range loudspeakers, for home theater you still need the subwoofers. The subwoofers give you the full force of the soundtrack and also protect your loudspeaker investment. That doesn't mean you can get a couple of mini-speakers and a subwoofer or two and that will do the job. People forget, or don't realize, that to get the best integration of a subwoofer, you have to have a full-range speaker to begin with -- a speaker that has good flat response at least one octave below the crossover point. 80Hz is a very common crossover point for subwoofers. With an 80Hz crossover point, the main speakers need to be flat out to 40Hz. You can do it differently, but there is more and more compromise the farther you get from the ideal.

We've been asked why we don't build a minimonitor. Quite frankly, bass is the foundation of music and I serve music, not equipment or people's ideas of what equipment they think they want. I haven't found a way to make a mini-speaker that goes down to 40Hz. I don't think mini-speakers are a legitimate way to reproduce music or to build a home theater around -- at least not at a quality level. People wouldn't be looking at Vandersteen loudspeakers unless they were looking for some level of sonic purity, and I don't know how to deliver that in a mini-speaker. Some critics think I'm naive on this point, but I think the market bears me out. If a small speaker could blend with a subwoofer, if that wasn't impossible to do well, wouldn't the market be dominated by little speakers with subwoofers? I wouldn't be messing around with 100- and 200-pound loudspeakers if I could make little lightweight speakers and subwoofers do the job. We already have what is renowned to be one of the world's best subwoofers, so that part is already taken care of. But the laws of physics don't play out that way.

dB: What are the newest Vandersteen products available today?

RV: Well, the Model 5 still feels new to us, but it's over a year old now. We have a video version of the 2W subwoofer, the V2W, for systems where there's a need for driving the subwoofer directly from the subwoofer outputs on a surround decoder or home-theater receiver. For serious home-theater systems and music systems we still recommend a pair of

2Wq or 2W subwoofers. Both continue in production. The 2Wq is sort of new. The Q of the subwoofer is adjustable, giving it the ability to sound like a music subwoofer for music and like a movie subwoofer for movies. We also have a 2Ce Signature and Vcc Signature center-channel speaker. The 2Ce Signature adds the tweeter and crossover parts from the 3A to the 2Ce. We also change the speaker-cable connections from the standard banana-plug sockets to screw-down barrier-strip terminals like we use on the 3A and 5, which are the best speaker-cable connections known to man in my opinion. The 2Ce Signature is $1500. The Vcc Signature has a different tweeter dome and crossover parts of the quality used in the Model 5. We go through a lengthy manual tuning process on each Vcc Signature. We basically customize the crossover to match the drivers in each Vcc. The individual tuning process is the same labor-intensive process we use with the Model 5. The result is that the Vcc Signature sound quality is an excellent match for the Model 5 and is even a step-up center-channel for a serious system using 3As and a pair of 2Ws. All the parts and manual labor it takes to create a Vcc Signature has a real impact on the price of the speaker. The standard Vcc retails for $495 while the Signature Vcc is $1099. We think the standard Vcc is fine for most people. But those with Model 5s will appreciate the close match in sound. So will people using 3As with subs if they are picky about their movie sound.

At this point we were running out of gas. A nice fresh-fish dinner in landlocked Hanford, California was a comfortable end to a long day of driving, touring the factory, listening, and interviewing.

One last stop to clean the incredible number of San Joaquin Valley bug smashes off the windshield and I was on the way back to San Francisco. On the way home it was interesting to reflect on the day with Richard Vandersteen. While we'd spoken before on the phone and at shows, this was the longest period of time we'd communicated face to face. He strikes me as one of audio's nice guys -- not angling for anything, just willing to talk honestly about what he thinks are the right ways to do loudspeakers and what led him to those conclusions. One thing is certain, he has optimized a lot of things on a lot of levels because he's been doing these similar and inter-related designs for so long. I was tempted to ask Richard for full details on each of the drivers in the 2Ce, 3A, 2W and 5 loudspeakers just so you could get an appreciation for just how special those drivers are compared to off-the-shelf drivers. However, we would have needed another 10 or 15 printed pages worth of space!

Years ago when the Vandersteen 2Ci was very popular, there was an insider joke going around:

"Why do Vandersteen speakers have those short legs on the stand?"

"I don't know. Why?"

"It's easier for them to walk out of the store when they have legs."

This alluded to how easy it was for dealers to sell the 2Ci compared to other high-end loudspeakers. Today that's still true for the 1C and 2Ce. Vandersteen loudspeakers are one of those crossover products, a high-end product that does very well in its arena, but that sells well to non-audiophile customers too. And that's just the way Richard Vandersteen likes it.

...END