31 New Home Theatre Products Presented

The Authoritative Journal Of The Widescreen Digital Surround Home Theatre Experience

WIDESCREEN REVIEW®

Polk SRT Speaker System Reviewed

H...

Au...

V...
E...

A Ne...

Vandersteen

Loudspeaker Designer Series
On Screen Interview By Gary Reber

Reprinted from Widescreen Review
Volume 5, Number 4, Issue 21

Loudspeaker Designers

Richard Vandersteen
Gayle Sanders
Ken Kriesel

DVD

Still Unresolved

Top 10 Questions About DVD

Exclusive Dolby® Digital Reviews

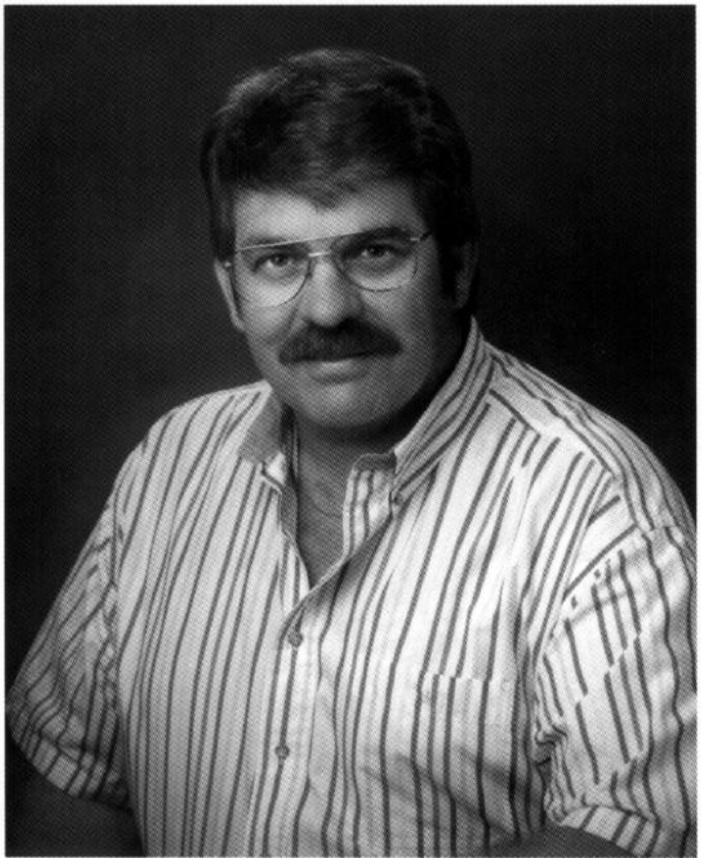
$4.95 US $6.50 Canada
Volume 5, Number 4, Issue 21

Richard Vandersteen

Loudspeaker Designer Series

Gary Reber

On Screen



Richard Vandersteen, Vandersteen Audio

Richard Vandersteen is the founder of Vandersteen Audio and the designer of one of the most respected and regarded brands of high-end loudspeakers in the world. Editor Gary Reber interviewed Mr. Vandersteen for this exclusive On Screen feature.

Gary Reber, Widescreen Review: One of the major things that separate the many speaker designs on the market today is radiation pattern. What are your views on point source, line source, dipole and bipole designs?

Richard Vandersteen: I'll take them in the way you have listed them. The first one-and I like the fact it's in first place because that's what we believe in-point source is the most correct in amplitude, time and phase, and I think this is the proper way to radiate energy into a residential space. The reason that I feel it is best is that I think time and phase are the most important parameters of how we hear something, like what makes an oboe sound like an oboe or a piece of glass breaking sound like a piece of glass breaking. Our ear/brain, as a combination, compares and integrates both time and phase as the most important parameters of what defines that sound. Amplitude is also important, but not to the degree that both time and phase are.

A line source can work well in a room because of the fact that the reflections off of all the boundaries in the room will be a pretty good facsimile of the original sound, so that the ear/brain, as a combination, can separate the direct arrival sound-which is its clue as to the true timbre or sound-from the room effects or the additions that the room or environment makes because the reflections are a facsimile of the original, however delayed in time.

Line sources have some very interesting possibilities. Their major problem in a residential room is that all line sources at some frequency become point sources. The energy from line sources and point sources falls at different rates as it travels through acoustic space. It is very difficult, unless you have a very definite known distance from the listener to the transducer, to calculate what your relative levels should be at that transition point where a line source becomes a point source. So, although very interesting, that's very difficult to implement in the real world, in a room that has a finite space. Of course, a line source in a very, very large room would behave as a point source and therefore, if the room were large enough, could still be satisfactory.

Dipole and bipole are similar and yet very different. Dipoles have a lot of disadvantages, They have all the disadvantages of a bipole, but they have one major advantage. That advantage is that most dipoles are ribbons or some sort of magnetic planer type or electrostatic type of driver. A lot of people tend to like that large radiating area-and what it really is, is an acoustic transformer from a large source into the low impedance of a room. The other advantage they have is these large diaphragms radiate as much energy from the rear of the diaphragm as the front, and the rear wave is not reflected back to the listener in a very short period of time by any large obstructions like magnets and baskets in dynamic drivers. There is no reflection until you hit the rear wall and of course that's some distance away. So they have this ability to be pretty pure in sound because there is no confusion from re-arrivals of energy that are reflected off the structure of the speaker itself.

Their disadvantage of course, is what do you do once this wave hits the rear wall? This complicates using a dipole in a real room. And they have cancellations of low frequencies which, by increasing the area of the diaphragm-that can be worked with -is another difficulty, especially in the bass,

I really don't see any advantage in bipolar designs. Because of reflected energy in all directions off of all room boundaries, they tend to throw a very big, large soundstage regardless of whether the program is big or not. This is popular, I think, because of the fact that it produces a diffuse and very large, pleasant stage and listening position. The problem is there is no way to get any real definition out of it because you have the direct energy coming directly from the speaker to the listener that is co-mingled with all of the same information, delayed in phase and time, coming from the boundaries of the room recombining on the way to the listener. So even if you have very high performance, very refined drivers with really good transient response and low distortion, that secondary radiation--if it isn't exactly a facsimile of the direct sound, which it can't be because of the difference in distance-is just basically perceived by the ear as a distortion. I really think that would be the least desirable of all the designs, although a very well known manufacturer realized that bouncing sound nine hundred and one different ways creates this type of sound that is very attractive to the consumers in the mass-market.

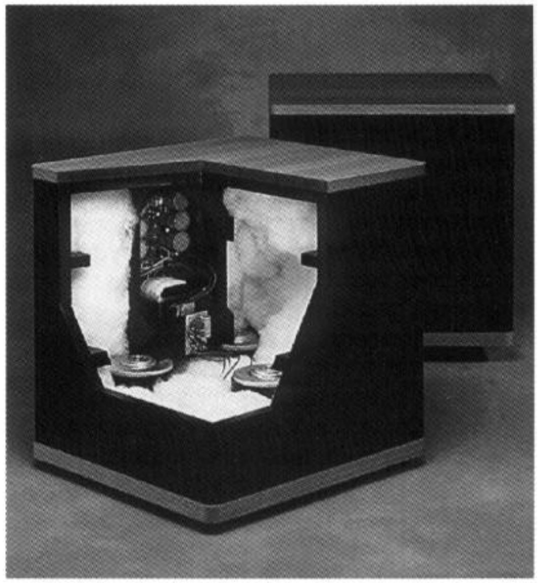
WSR Reber: How these different designs interface with the room also differentiates them. Are your speakers designed to be used at or near the room boundaries, away from the room boundaries, on the floor or stand mounted?

Vandersteen: We make products that are designed to work in different situations, We make subwoofers which are designed to be in or near a corner, because in the practical world that's where they are going to be put. So we "Q" our woofers at -5, which is considered to be lean for video purposes, but when you put them into the corner, the effect of room gain bumps things up to what I believe is a proper "Q" for woofers, maximally flat .7. There is some disagreement on that, but there is more agreement than disagreement. Also in the real world, the center channel speaker and the rear channels in most environments are going to have to be near the room boundaries for space or cosmetic purposes, The center channel for sure is going to have to be either above or below the TV. Our center channel in fact has a proximity switch so you can adjust it for just how large that immediate boundary is, because it does have a dramatic impact on the frequency response, especially in the low frequencies. Our main speakers are designed to be in free space, some distance away from the side wall and rear wall-and that distance should never be the same. You don't want those distances to be the same, you want them to be a non-multiple of one another. You get natural openness that way, and that is the ideal way to place a loudspeaker. So I guess to answer your question, ideally you would have all free-standing loudspeakers, but sometimes you can't. We make an on-wall for rear channel video use, or for aesthetics Some people with small rooms have to mount their speakers in a bookcase or on a wall, and it's designed exactly for that purpose. All of our products are phase, time and amplitude correct in the environment in which they are designed to be used.

WSR Reber: Richard, why don't you describe for our readers in layman's terms what you mean by phase. Just so they can get your view point on this often used term.

Vandersteen: Phase is simply the relationship of positive and negative pressure waves as seen by the microphone during the recording. When there was a positive percussive phenomenon in the recorded space, it was perceived as a positive pressure by the microphone. Modern electronics preserve that quite well, except when extreme equalization is used, but in the high-end world equalizers are not that common. We are working and dealing with high-end people in the high-end part of the industry. Extreme equalization or even moderate equalization can cause severe phase shift, so everything I say would not be relevant through an equalizer. But knowing that our industry doesn't use equalizers, I believe that it's very important that this phase be preserved. In other words, these positive and negative poles must be preserved throughout the amplification chain and the wires and interconnecting cables, but also, when the signal finally gets to the transducer to be put into the room-to be radiated into the room, that all of the woofers, tweeters and mid-rangers, that all of the diaphragms involved in the drivers, should be moving in the same direction at the same time. This is especially important in two-channel, but it makes a difference no matter how many channels are used.

"We make subwoofers which are designed to be in or near a corner, because in the practical world that's where they are going to be put."



Vandersteen 2W Subwoofer

There are a lot of speakers that have the midrange wired out of phase electrically from the woofer and tweeter. Out of phase means the reversal of a positive going force turning it into a negative force going into the room. And it's audible to most people. Can you imagine then how audible it is when the whole speaker is phase correct? Our eardrums, and most things in nature, do not respond the same in both positive and negative directions. I believe because of that, if you have a speaker going in different directions at different frequencies, it's not as linear because we hear it in a non linear way. And I believe the entire spectrum should be then preserved in phase. People accuse me sometimes of being too sensitive, or maybe over reactive to phase, but take one of your stereo speakers and wire it out of phase. Now remember this when you do that: the information you hear is still identical, nothing has changed, the amplification, the source, everything is identical, all we have changed when we do that is phase. The speaker is the same, everything is the same. The only thing that has changed is the phase to one channel. Listen to the big difference.

WSR Reber: How would you relate to our readers time and amplitude?

Vandersteen: Amplitude is, of course, how loud all of these different frequencies are. It's just a measurement of energy in the room. Phase and time are very, very interrelated, you can't have one without the other-if you have good phase response, you will have good time response. It's impossible to preserve a nice square wave or a waveform that would be a facsimile of let's say a note of music without accurate time and phase response. If you wanted to truly pass that waveform on as the instrument put into the microphone, you would have to have the timing of all of your drivers, the tweeter, the midrange and the woofer, such that, as the signal from all three of those drivers convene to make that one waveform, each driver responsible for a different part of that one waveform must be in phase. If you want to see that, the way it originally was captured by the microphone, the speaker must be time synchronized. It also must be phase correct, or it would cause part of that waveform to reverse into the negative direction and it would no longer be a copy of that original signal if you were to look at it on an oscilloscope. Any deviation from that original signal-and nothing is perfect mind you-is a distortion.

WSR Reber: Time and phase go hand and hand basically is what your saying?

Vandersteen: Absolutely.

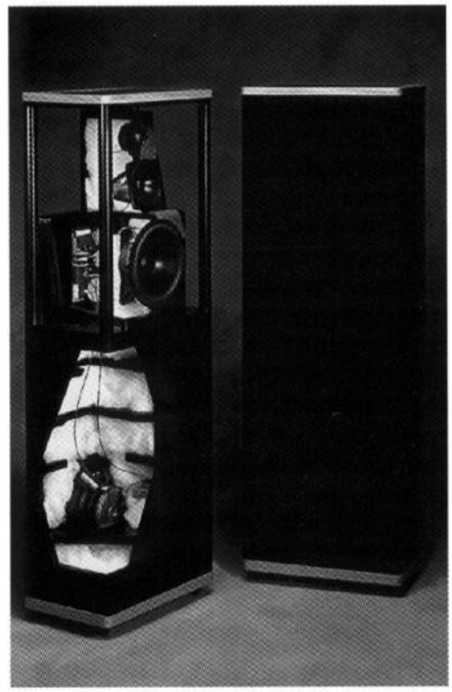
WSR Reber: Should speakers used in a multichannel array be voiced differently than stereo speakers, for example, flat vs. a warm tonal balance?

Vandersteen: I think that any "voicing" or altering of the signal should not be the job of the transducer, whether we're talking about sound for audio, video, or even a purpose that hasn't been defined yet. Transducers in any multichannel system, two-channel or even a one-channel system, should offer an accurate reflection of the signal fed to them. For instance, if it's not warm enough, that's the function of a tone control, to cut the highs a little bit. If there's not enough bass, a tone control can slightly boost the bass in order to alter the signal. The function of the speakers in any system, I call them transducers, is to accurately put into the room, the signal that's given to them. I am amazed by the "thumper bumpers," the woofers that we use in video, because there is a demand for very heavy, very strong bass in video. From an artist's point of view, I have no problem with that, I kind of like it too, but it shouldn't be part of the design of the woofers. The woofers should be designed to be accurate and some sort of tonal modification in the system or at the preamp level or processor level, should be available for the consumer to go ahead and boost it if he or she wants more than what's actually in the program material. Then you won't have this big discrepancy between theatre systems and music systems because the system could be accurate, which means it will do music very well. And when you go to play a movie, if you want more bass for the impact on dramatic effects and everything, well then simply do it the proper way, turn the bass control up. So to answer your question, I believe the speaker should be flat. Warm tonal balance-if that's what's desired-that's the domain of tone controls.

"I think that any 'voicing' or altering of the signal should not be the job of the transducer..."

WSR Reber: Do you subscribe to "timbre matching" of the rear speakers, the idea that a rear speaker should be voiced differently than its opposite in front, as THX states, to counteract the different head related transfer function the listener experiences when sounds arrive from the side and rear?

Vandersteen: Well, "timbre matching," I don't know what they really meant by that, but I describe timbre as a relationship of amplitude, phase and time. And all of the speakers in a surround sound, THX or any system should have the same time and phase response and amplitude response when they are used in the locations where they will be. In other words, the center channel under or above the TV set, a pair of main speakers out in open space, and generally speaking, if they have the room and the room is large enough, another set of free-standing speakers for the rear channels. If the room isn't big enough, and if that means that a free standing speaker would need to be crowded up against the wall, you are much better off with a speaker that has the same time and phase and amplitude response, but is designed to do that against the wall. If you take an identical speaker to what is used in the front and use that inappropriately, and put it near the wall in the rear, you will not have the same phase and time and amplitude response. THX also talks about



Vandersteen Model 3A

correlation or decorrelation, which is simply a function of mono in the middle, it's an intercranial kind of experience that occurs when you're exactly the same distance from two mono sources. The rear channel of Pro Logic is mono, and you can decorrelate it by just having a sweetheart sit next to you and move yourself about twelve inches, and you now are not equal distance from the rear speakers anymore and the problem goes away.

WSR Reber: I like that sweetheart idea. We'll come back to more of this discussion. Do you favor room treatments to alter acoustics? If so, what type?

Vandersteen: I think that some room treatments can be very, very beneficial. For instance, if the listening position needs to be very near the rear wall, it's wise to put some diffraction or absorption material directly behind the listener so that the reflections off of the rear wall are modified. We need a little bit of time for the ear/brain to hear things the way they were meant to be in the recording. Likewise, if your free standing speakers need to be near the side walls-if you can imagine a cue ball, if you were sitting at the listening position and that's where your cue ball was, and you were to shoot it off of the side wall so it would rebound and hit your left speaker, that point at which that pool ball would have to bounce off of the side wall-it can be very beneficial to put some diffraction or absorption at that location. Generally speaking though, when speakers are properly designed in phase and time, and are not excessively bright, this is not necessary. The neat thing about time synchronized speakers is when you stand up, you are automatically out of alignment and the high frequencies are attenuated so you don't have a lot of problem with ceiling reflections. The floor is usually covered with carpet which is good, but excessive absorption on the side walls just compresses dynamic range, causing a person to drive his or her system much harder, with much more power, and you still get kind of squashed, compressed sound with more distortion. Obviously if you have an empty room with a lot of echo in it, you need to deal with that and I believe you should deal with it until you have a reasonably even decay of amplitude and then do what you need to, but don't go to this totally live-end, dead-end, or dead-live-dead like some people now are doing. I think it just compresses dynamics.

WSR Reber: How about electronic equalization?

Vanderstaen: Electronic equalization, if you are into phase, time and amplitude, is not possible.

WSR Reber: Even with the best room equalizers?

Vandersteen: Even with the best equalizers. Any time you do equalization you are shifting time and affecting phase. There are exceptions though. Our new Model 5 loudspeaker system uses very sophisticated equalization. We modeled all of the potential standing waves in a room, using a very broad variety of different room dimensions. We ended up with a family of most likely scenarios. At thirteen different points below 120 cycles, from 20Hz to 120Hz. In the Model 5, the equalizer acts on the bass amplifier only and does not effect the signal going to the mids and highs or to the mid-base for that matter. Because once you get to the point where the wavelengths of the frequencies being reproduced are larger than the dimensions of the room, the ear is no longer adequate to separate the reflected energy from the direct arrival because it is all basically in one time frame, and I believe that some equalization to correct for low frequencies can be very beneficial and we are implementing that in the subwoofer module of the Model 5. Anywhere from 120Hz to 140Hz on up remember the speaker is supposed to function in place of that trumpet, that glass breaking, that phone ringing, that woman talking, that guitar or whatever it's trying to reproduce. What if you were to equalize that range because the room acoustics might suggest that? You run an average response curve of what a speaker is doing in a room, then you equalize to flatten it out. Now let's imagine you took a Steinway and put it in that same imperfect room. Would you go into that Steinway with popsicle sticks and cotton balls and modify the tonal balance to try to make it "more flat" in that room? Then it would sound like a screwed up piano in a screwed up room.

"Any time you do equalization you are shifting time and affecting phase."

Once you have enough distance from the reflecting boundary for the ear/brain to separate the direct sound from the reflected sound, then that original signal must remain an accurate facsimile of what it is trying to reproduce, in my opinion, and let the room do to it what it will. Because the ear and the brain can separate that, you will understand that. You can experiment with this by doing heavy equalization in the midrange. It just sounds like a very strange instrument. Just because the average energy in the listening position is now flat, it is not perfect. Remember, part of it is contributed by the instrument through the loudspeaker, and part of it is contributed by the room. A flat average response does not make it sound like that instrument was in that room. It sounds like there was a sick instrument in that room.

WSR Reber: Do you believe that what's right for stereo is right for multichannel in the context of the effect of speaker directivity on the precision of imaging when more than two channels are present?

Vandersteen: Yes, I believe that accurate transducers are needed for any number of channels. The only unique thing would be maybe the center channel.

WSR Reber: There is a lot of discussion about bass loading methods. Do you favor second order sealed boxes, fourth order vented, or passive radiator designs, band pass, dipole, bipole or other methods?

Vandersteen: My preferred method of doing bass is second ordered sealed boxes, although when you design them with a very low "Q" they actually approach first order. They can get large or they need to be powered in order to accomplish that properly. But I believe the pitch definition and the real world transient response-all things considered-is better sounding. Passive radiators or vented are one in the same. Either a passive radiator or a port can give you lower frequency response out of the same size box, however, in the time domain, the reverberation pattern, or the transient response will be negatively impacted to one degree or another. Using a dipole or bipole for bass is not practical. There are just to many cancellations and everything involved there. Of course it would be ideal if you had nice neighbors, you could just pop a couple of woofers with the electrical-mechanical combination "Q" of about .5 to .7 and pop them right in the walls and let the outdoors be your infinite baffle, that would be of course the best of both worlds, but not practical for a lot of people. You would have to remember that your home would then act as the box, and your neighbor would be the listener.

WSR Reber: I want one of those. (laughter)

Vandersteen: That would be ideal. In the real world, that would not be possible. It is what I use at home, but I have neighbors to consider.

WSR Reber: Now, you can not say that "in the real world that it wouldn't be possible" if that's what you do.

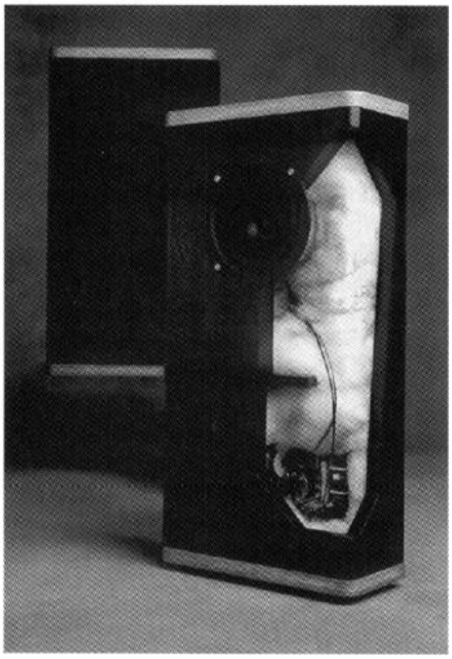
Vandersteen: What I mean to say is that it is just not practical for many people, but it is what I use.

WSR Reber: So you do that at home?

Vandersteen: Yes. But we can't listen to it all the time because we do have neighbors. It is important that the driver to be used for that have a combination of electrical and mechanical "Q" near what is considered optimum, .5 to .7.

WSR Reber: Explain "second order" for the readers.

Vandersteen: Well second order simply means that the speaker acts as a high pass filter. Once the driver goes through resonance, the output will fall at 12dB per octave. Passive radiators and fourth order designs fall at between 18 and 24dB per octave. Now in the real world, the steeper the roll off, whether it is a passive elec-



Vandersteen VSM-1

trical filter or an active filter in a piece of electronics; or whether it's a loudspeaker, the steeper the filter the more ringing you are going to get out of the filter, because they all function in the same way. If you use a mathematical model, they wouldn't look different: The higher the order of roll off, the more ringing you are going to get. Obviously, going back to what we said earlier where a speaker should reproduce what it's fed, if it's given a pulse and you were to measure what the woofer puts into the room, it will give you a current facsimile of that pulse, but the driver wouldn't stop there, it would reverberate on. That is called ringing. Obviously that was not part of the original signal, so that would be a distortion of the original signal. You have more of that energy outside of the original pulse the steeper you go with the roll off. Steeper filters mean more ringing and more distortion.

WSR Reber: What are your other views on subwoofers that you want to express?

Vandersteen: Well, I think subwoofers are important. The interesting thing about subwoofers that isn't very widely understood, is that a subwoofer is usually a separate box including some drivers that are very capable of doing very low frequencies with high output that would then have to be crossed over into a full range speaker or a speaker that's going to carry the midbase, mid-range and high-end. One thing that you see that is very common today, are three piece systems where you have very small, little mini monitors and then one or two subwoofers. And that's a very very difficult thing to achieve and it can be argued that it can't be achieved. Because you learn in "Filter Theory 101," on the second day in loudspeaker design school, that there's one very strong consideration in order to accomplish a good crossover: you need to have good flat response one octave above and one octave below the crossover point. If we pick 80Hz as a reasonable crossover point, that means that the mini-monitor speaker or the main speakers that are going to contribute the midbase and highs need to have good linear response down to 40Hz. We don't make a mini speaker because we don't know how to make one flat and linear down to 40Hz. So in the transition from the main speaker to the subwoofer, if you require this one octave above and one octave below in order to accomplish that transition correctly, what you're basically saying is in order to properly use subwoofers you have to start off with full range main speakers and I believe that's very easy to demonstrate. So having said that, that means that the subwoofer needs to have good cone control in order to have good linearity an octave above 80Hz, which takes you on up to lets say 200,160 to 200Hz. That demands a lot of those drivers that are being used because at that point you could get a lot of non-linear, secondary radiation out of the subwoofer, and that needs to be controlled. That's why I believe that bottom firing, slot loaded, multiple small drivers is the most successful, practical way to properly do low frequencies in a home environment. And that is what we do.

"...I do believe that stereo subwoofers are a very viable option, placed at the left and right front."

WSR Reber: The views you just expressed are quite contrary to the approach the loudspeaker designers have taken to get THX certification.

Vandersteen: I understand that, but there is a very good reason why we do not have our products THX certified. Although we could certainly design products to meet those standards, I just believe they are not high standards.

WSR Reber: Do you recommend a dedicated subwoofer to reproduce the ".1" low frequency effects or LFE channel for maximum impact?

Vandersteen: Well, that's a very interesting thing in that one might use the .1 to run a chair shaker or something for effect to give maximum impact, if that is what the listener was really looking for, but in all the AC-3 processors I've experimented with-and I don't understand the phenomenon here-but all of them seem to sound better when you don't use the .1 out. I know that this automatically redirects the .1 energy into the other channels, and I am at a loss really to understand why, but every processor that I've played with, which is about six or seven of them, sounds better with the .1 not in use. In other words, redirecting that energy to the other channels and using subwoofers on one, two, three, four, or five of the main channels sounds better. So I guess in the home, unless you were trying to drive a fixed kind of machine like a chair shaker, or something to pound on the floor joists, I guess I would say that the LFE should be left alone until you have five subwoofers in the room. Then I would guess that there would be an argument for adding the sixth one.

WSR Reber: In a couple of my reference systems that's the way they are configured and even then the ".1" is intermittent as it's used in a soundtrack.

Vandersteen: But, that's energy. The processors that I've played with anyway, when the .1 energy is there, it is basically by definition .1, it is one, therefore it is mono and it is interjected in equal amounts to the five channels. Why that seems to sound better, I don't know. I sometimes wonder if in the Dolby specification there is some very, very low frequency pole put in the main discrete channels, when the .1 gets used, or that mode in the processor is selected. I sometimes wonder if there isn't another pole or something engaged in the process, because it would explain that kind of ringing effect. The deterioration of the sound that I hear is very similar to going from let's say, a first order filter and going up to maybe an 18 to 20dB per octave filter. There is a time domain disturbance of some kind going on. I have not investigated this, it is just an observation and it is why in my own personal systems I am running five subwoofers on all five channels and not using the .1 out.

WSR Reber: That possibly could explain why I have noticed that the THX trailer logo in the PCM format which would be decoded by a Dolby ProLogic decoder with no ".1" output, sounds much more coherent and substantial and powerful in the bass. When you play the logo off the Dolby Digital track in which you have a ".1" subwoofer connected to the system you have this very weak bass response and non coherent sense of sound.

Vandersteen: I think some of the processors when they redirect this .1 energy, some of them redirect it but at a lower level. This of course could be compensated for by just turning the woofer levels up on all five of your woofers on the other channels, because it is a mono source, that means it's injection into the other five channels would be exact copies of one another. So even if you were only running one subwoofer, lets say off a left or right channel, you could in effect compensate for that by just turning it up when you do your calibration.

WSR Reber: Right.

Vandersteen: I don't recommend one subwoofer in any case. But I do believe that stereo subwoofers are a very viable option, placed at the left and right front. With the LFE redirected, you could compensate for that by just running your subwoofer levels a little higher even if your rear channels, because of practical considerations in your installation at home, were limited in low frequency response.

WSR Reber: Well, if you wanted to optimize a 5.1 channel low frequency bass system would you recommend subwoofers at all five plus the ".1" channel?

Vandersteen: No. With my experience at this point I would recommend using 5 subwoofers on all five main channels and redirect the bass from the LFE or .1 channel to the other channels.

WSR Reber: Do you think that bi-wiring offers an audible improvement that justifies the cost? Can your speakers be easily bi-wired?

Vandersteen: Bi-wiring is very interesting. We have been doing it now for twenty years. Our smallest model, the 1C is not bi-wired, and we're often asked why is our littlest speaker not bi-wired. One of the main advantages of the bi-wiring is the separation of the magnetic fields in the wire. I'm talking about the expanding or collapsing magnetic fields that occur when an amplifier is trying to drive a wire connected to a very large, heavy duty woofer. This results in a tremendous amount of current going to that woofer and because it is not a simple resistive load, it's a reactive load, the back EMF can be very significant, and you have to remember expanding and collapsing magnetic fields is how transformers work, so it can be very significant if that field is crossing through the very same wire that is driving the subtle signals involved with the midrange and tweeter. The reason we don't do it on a small two-way is because it's a very light weight responsive 8-inch woofer. So when you are talking about small, little speakers, I think bi-wiring just makes it sound different. It can't really be justified from an engineering standpoint and the money it costs to do it would be better spent on higher quality components for the crossover or better drivers. On the other hand, when you have a full range speaker which has a large low frequency driver in it-you know a very heavy duty, long-throw eight, ten or twelve-inch or anything larger-then being able to separate the magnetic fields by the use of bi-wiring can be nearly as big a sonic improvement as bi-amping. I sometimes wonder if in the early days, the tremendous improvement that we got when we bi-amped things, 60 to 70 percent of what we heard was caused by bi-wiring and had nothing really to do with using two amps. Modern amplifiers today can handle 20Hz to 20kHz just fine, and a lot of the improvements that we hear when we bi-amp may be because we also bi-wired at the same time. It's pretty common nowadays though to have these bi-wire schemes where you have the wires within one jacket. This of course doesn't take advantage of the main value of bi-wiring, and therefore is not really bi-wiring, it is a sophisticated way to mono wire. I believe a few of the advantages of bi-wiring are still there, so it probably should be described as the ideal way to mono wire a bi-wired speaker. However, it is not true bi-wiring. True bi-wiring can only be accomplished when the leads carrying the lows, vs. the leads carrying the mids and highs, are separated from one another about an inch or two so that those magnetic fields can't couple their energy into the leads.

You have to remember that a bi-wired system with the leads separated an inch or two reflects the impedance of the crossover all the way to the amplifier terminals So the transition of the cross over of the lows vs. the highs is actually occurring at those junctions at the amplifier terminals, because the wire attached to the midrange and tweeter will represent a very high impedance at let's say 20Hz. That signal at 20Hz is going to take the path of least resistance, which at that point would be the wires that are hooked up to the woofer, because it represents a 4ohm or an 8ohm load. The wire that is hooked up to the midrange and tweeter at 20Hz is going to represent about 100ohm load. This is how crossovers work by reflected impedance, so that is just like physically removing the crossover from the loudspeaker and putting it at the amplifier terminals. I think one of the reasons why there is so much controversy about bi-wiring is that there aren't many who truly understand how it works. There are a lot of people doing it just because people are doing it.

WSR Reber: You certainly believe that time and phase are audible parameters in loudspeakers. Do you favor first order crossover slopes or steeper slope designs? Do you physically position for time synchronization the drive elements in your designs? Is diffraction important? What if anything do your designs use to affect diffraction?

Vandersteen: Yes time and phase, of course, I think are the most audible parameters of a loudspeaker, and that's why our speakers have correct time and phase. If you believe in time and phase accuracy and you want to pass a pulse as it was originally captured by the microphone and passed on to the speaker by the amplifier then you have no choice but to use first order slopes because those are the only

"We do physically time synchronize the elements in our designs because that is really the only way to do it."

ones that will preserve the phase integrity of the signal that came from the amplifier in the first place. First order crossovers are extremely difficult to do, and it's very demanding on the drivers. The quality of the drivers you use has to be very high, and they need to be very rugged and durable. It is kind of something that I would rather not have to deal with, but it's the only way to preserve the correct time and phase. We do physically time synchronize the elements in our designs because that is really the only way to do it. To align the arrival times of different drivers if they weren't physically staggered is virtually an impossible thing to do in a passive crossover. It requires an incredible number of coils and they need to be designed to very, very, very tight tolerances, and they tend to ring and I've never seen a practical application where it was really done. Many people have implied that they've done it, but I've never seen it. We've experimented with that, and let's just say that for all practicable purposes it can't be put into production. So the only way to time sync the drivers is to physically stagger their positions. This would simply mean that a woofer has to be forward of the tweeter so that the energy from each arrives at your ear some distance away, at the same time. The only way to accomplish that if they were all on the same plane would be to lean the speaker back a lot and then you can't really get them all right. Or you could put a separate amplifier on each of those drivers and then do the time synchronizing in the digital domain or with bucket brigades-you know, the way it was done in years past.

WSR Reber: Explain to our readers what a bucket brigade is and define first order.

Vandersteen: Bucket brigades were before we had digital delays like today. It was an electronic circuit that could hold a signal long enough to be re-released with some period of time in between. That's all it was. It's an old fashioned method of delaying the energy and releasing at a different time.

First order is where the acoustic output of a driver and the electrical characteristics of a crossover are manipulated to fall at 6dB per octave in energy when they roll off. It's not so important what the driver and the crossover are doing separately. What you want, for at least one octave anyway, is for the combination of the crossover and the driver to combine to produce an acoustical roll off at 6dB per octave, or first order.

WSR Reber: Is diffraction important?

Vandersteen: Diffraction can have a very, very negative impact on the sound of the loudspeaker. Diffraction is caused by having a large baffle around drivers or having edges or any structures that haven't been rounded or contoured in such a way that they splay their reflected energy so that it randomizes and doesn't focus at the listener excessively. The effect of diffraction can be demonstrated by talking through a piece of cardboard with a hole in it. Have a person you know talk through that and you will see how severely their voice is altered by this baffle board. So we all strive to have high quality, accurate tweeters and midranges, but the minute they are mounted into a baffle that accuracy is lost and manipulated, and the reason for that is a good percentage of the energy that is radiated by the driver is actually reflected back off of the baffle board and back to the listener. Now this is not only delayed, but also shifted in time and phase to a degree. It may be subtle, but it can be measured to be quite a distortion. The signal is no longer a facsimile of what was sent to that transducer.

"Diffraction can have a very... negative impact on the sound of the loudspeaker."

You can round off the edges of a baffle board or make the baffles narrow and rounded off. All of these things control diffraction and minimize the negative effects. Of course the best thing to do in our opinion is simply just remove the effect of the baffle all together by not having the baffle. My designs have done that from the beginning in 1973. Inside, our speakers look like a whole bunch of little boxes on top of one another, with the face of the baffle around each driver just barely large enough to hold the driver that it is supporting. And of course there still needs to be very careful attention paid to the edges of that structure with rounding and so forth. The structure that supports the grill cloth should be some distance away and then of course round again to keep reflections from focusing. Sharp corners tend to focus these negative things and spot light them and you could minimize that by rounding them. So what we have done is basically removed the enclosure. We built a boxless speaker where we don't have the reflective baffles in the immediate proximity in the drivers and the structure is held far away and at a very large angle away from the listeners access so that the structure that is needed to make it work is not audible.

WSR Reber: In addition to test instruments what source material do you use to evaluate loudspeakers? Music? What do you recommend our readers use? How should they listen?

Vandersteen: Of course test instruments are a tool to help the designer. Actually what they are is a short cut so that there is not so much cut and try. All of our evaluation is done on music. It became clear to me many years ago, that if I was really going to know what I was doing from the design stand point of loudspeakers, I needed to have my own master tapes. And as a result we have made a few of our own recordings so that I have reference material and know what it really sounds like. Prior to that we were using various recordings, commercial recordings, and some state of the art recordings from our industry, but you never were certain, you never really had an exact idea of what that recording really sounded like when it was made. As a result we went into that incredible expense and went out and did a few very high quality reference recordings. They are available through Vandersteen Audio dealers, however, the real purpose was not to be sold but to use for evaluating our designs. The ultimate defining thing we use for evaluation is music, but we've done a lot of live versus testing. Things like scraping shovels on the concrete, shaking car keys, any unique noise that can be recorded or be played back in real time to see whether a given design is doing that accurately or not. That's by and large how we do nearly all of our evaluation of different designs. We take it as far as we can with the instrumentation and then we do it with live versus testing.

Consumers do not have the advantage of their own recordings in most cases. They should choose simple recordings for reference-like jazz, female vocals, things that are not real complex tend to be recorded more simply. Then listen over a period of time, evaluating a given system, speaker or whatever it is that a person is involved in, for its ability to sound like the piano, the guitar, or whatever is real. Does it remind you of live music or what those instruments or voices sound like in real life?

WSR Reber: What is your view on the suitability of home theatre systems for music reproduction and vise versa?

Vandersteen: Well, I believe that our home theatre system is just as viable for music as a dedicated music system because it is based on a music system and it uses all the same components, just more of them. I believe as time goes by that people will see, and I have done experiments already at this point, that the basic matrix system that was done by Dynaco and Haffler and people like that, properly adjusted, can already render two-channel for music listening as not as rewarding an experience. A multichannel system with that type of decoding takes advantage of the natural ambient information that is already in all two-track recordings, I believe that when our processors start taking that simple matrix more seriously and start applying digital delay that we'll really have something. One of the reasons the passive matrix systems were not a run-away success for music listening was you tended in most situations to be nearer to the rear speakers than the front, and you had this eerie effect of the effects hitting you before the direct sound or the real happening from the front speakers. With high quality delays you can now delay that rear signal so that is not an issue anymore. Using those principles and adjusting them to a level where they are not exaggerated, you can literally draw the image out of the pair of stereo speakers to the point to where if you turn the center channel and the two rear channels off it sounds like something failed in the system.

And of course the way to adjust these systems correctly is to adjust the center and rear speakers so that they appear not to be doing anything and you notice how much they are doing when you turn them off.

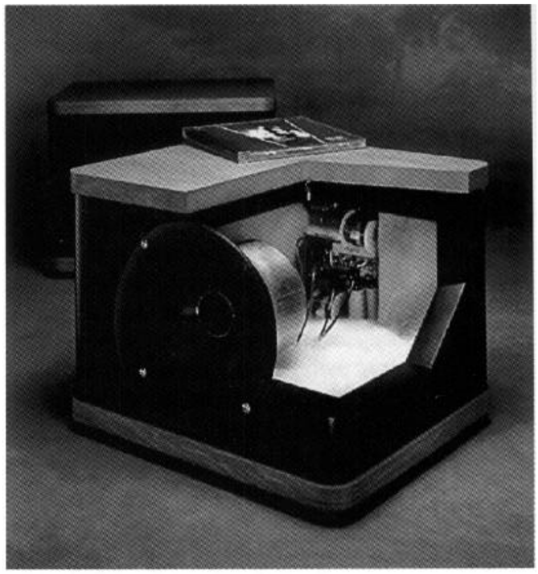
Theatre systems as they are called, or THX for instance, of course that's an abomination on music, is I really believe as much an abomination on video sound as it is on music. Movies are filled with music. They're filled with events that we experience everyday. Phones ringing, dishes clanging against one another, things dropping on tables. It's amazing to me sometimes how accurate the sound effects are in movies and I know there is no reality there. I know it is all done separately, but those effects were all created by someone clapping two things together-whatever, and the more accurate the loudspeakers, in my opinion, the more accurate those sounds will be.

The typical theatre with the equalization that they do and the obnoxious amount of low frequencies that they put in them and everything, and the very, very forward harsh effect that the speakers tend to have because of the way they must be designed all just ruins part of the effect for me. It's just not as natural because it is not as realistic I think ultimately in the end, all of the theatre systems will come down to that because the people get tired of being impressed solely by how loud the system is and how boomy it is. At some point people mature, or they have guests over and especially if females are present they are going to have to turn the levels down in order to have the gals stay with them because it physically pains them, they can't handle it. Then if all there was as a basis for the system was volume and boom, then what is left of the naturalness and accuracy of all of the sound effects and the music? There is where the reality is.

WSR Reber: Do you advocate using identical speakers at each position in a multichannel system? I am framing that question in terms of optimum approaches.

Vandersteen: Yes, except for the center channel because it's always going to be either under or above a TV set. If you have a very large room where you could be far enough away from a pair of our main speakers like the Model 1,2,3 or 5-you need to be some distance away to give the drivers all a chance to converge and become coherent-then four identical speakers at the four corners can work. However, if those rear channels need to be nearby the listener, like at the ends of a couch, or if they need to be near the side or rear wall, as compared to the fronts, then I believe a special purpose speaker designed to have the same size, phase, time and amplitude accuracy in that hostile environment is by far the best way to go. And that's why we designed the Vandersteen Audio VSM-I speakers for the rear channel. They maintain accurate phase, time and amplitude response in the real world. So in a sense, that's saying that you should have the same speaker all the way around but if you can't do that, then at least have speakers that sound the same when they are installed the way they are going to have to be.

WSR Reber: This is another variation of that question. Do you recommend identical time-distance relationships of each speaker from the sweet spot listening position in a multichannel system?



Vandersteen UCC Center Speaker

Vandersteen: Yes, except for the center channel. The center channel always appears to fold into that stage that is presented by the front channels if it is delayed. So, if you can, I believe that pulling the front speakers a little bit towards the listener so that the center channel is further away from the listener sounds more realistic. It's just more coherent in the soundfield that it creates across the front of the room. Rear channels tend to sound better in my experience when you can get them up a little bit, a little higher than ear height. Of course then you need to be careful that the time and phase characteristics of the speaker are preserved when they are installed that way. So in a sense, I think that the question you are asking here, is time important, and the answer to that question is yes.

WSR Reber: Do you see time implementation of that distance relationship better achieved through physical location of identical feet distances of the speakers to the listener sweetspot or in terms of electronic delay time distance compensation.

Vandersteen: Well, obviously if you can do it physically that will always be the lowest distortion way to do it. But in the real world most people can't accomplish that. And then again, second best would be to do it electronically.

WSR Reber: How do you view speaker placement to optimize phantoms between left front and left back speakers, and conversely right front and right back, or for that matter, left back and right back? And what about along the diagonals between left front and right back and right front and left back?

Vandersteen: Well, here I am going to disagree a lot with convention. When I have been in a situation where I have had my choice, I have always had much greater success putting the rear channel speakers on the rear wall, directly opposite of the front channel speakers up front. The sidewall placement that is common in many systems, is less effective in my view. If you use those fuzz boxes, those dipolar fuzzy things that have no definition anyway, then I don't think it matters much. But with direct radiators and especially on AC-3 or DTS* where you have discrete signals, I have much greater success putting the effects channels on the rear wall. The other advantage that has, is it takes the listener out of the direct on-axis response of the tweeters, which can be very beneficial at the rear of the room because they are usually going to be pretty close to them. Now when you do that, if the program material has it in there, and *True Lies* has some very interesting places in it for instance, then you can literally get imaging on the ceiling, phantoms created on the ceiling, down the side walls. So I think that as they continue to perfect discrete technology and as they perfect how they record these things and make movies, playback systems may evolve some.

"...I think that as they continue to perfect discrete technology...playback systems may evolve some."

One of the things that I have tried with my system, because left and right in Pro Logic tends to be steered so hard, is to leave my left and right channels full range, with no processing. I don't run them through the processor at all. I take the second output into my processor and I use the center out going to the center channel and I adjust it 4.5dB down. And then what I do is I pull strings from my main speakers to the side wall and I mount a pair of our on-wall speakers there, and that's where I put a left and right channel out of the processor and then I put left and right rear on the rear wall pointing forward and then the gain of the whole thing goes up and down so you have to do it at pretty high gain otherwise your signal-to-noise suffers in the processor. But I find that to be a very, very interesting affect where you can literally draw the effects of the front channel out to the sides. The stage that is being presented in the front of you is way, way beyond the sides of the wall. I just really love playing with this stuff.

WSR Reber: I prefer the physical time distance relationships to optimize my systems and I basically draw a huge circle from the sweet spot from the center of the room and then, after I set up my front left, center, and right speaker positions on this time arc I hook up left front and left back as a stereo pair and from the sweet spot position I actually play stereo material to optimize the relative positioin of the left surround with the left front for the best possible phantom imaging perspective. I do this for each side wall and the back wall to fine tune the phantom imaging perspectives. I do physically delay my center by having it physically behind the arc on which is positioned the left and right front speakers. The terms I use to describe my approach are Symmetrical Holophonic Imaging Array™ and Equidistant Speaker Time Synchronization™.

Vandersteen: Yes, because you would have the pseudo stereo across the rear then. The end result is fantastic, especially when the recording engineer and artist uses the five channel discrete palette to really create a holophonic experience.

Vandersteen: Exactly.

WSR Reber: It is how they mix it is how it is going to come out.

Vandersteen: Yes, of course, if you have the physical room to accomplish what you are doing like I said earlier, being able to do it in physical placement is the lowest distortion, most preferred way to do it. If you're not able to do that then doing it electrically is the second best. But mine is good. So even if you can put yourself in a big circle with your five channels surrounding you, I still think the center channel needs to be delayed physically if possible.

WSR Reber: I agree with you there. I think it integrates much better into the front pair for a more coherent front hemisphere presentation.

Vandersteen: I think that is somehow related to the precedence effect and I really haven't studied the phenomenon. It's really not my area of expertise, it just works-that's why I do it.

WSR Rebar: Should speakers in a multichannel system have identical full range capabilities for each discrete channel?

Vandersteen: Well, ideally yes. If that can't be accomplished then it is most important that the left and right front be full range, and that the rear speakers get down below I00Hz, because that is where most of your ability to locate direction lies. Most of the frequencies that are locatable by the ear are I00Hz and above, so that certainly needs to be handled as discretely as possible. I believe if one can afford subwoofers that it would be best to have five of them in the room, and in the case of an on-wall speaker, it should be flat down to 70Hz or so-and it can be because of the predictability of the wall-to give you one octave of linearity that you would need for even transfer into a subwoofer. So ultimately yes, you would want all channels full range. Now, if that's accomplished with five identical full range speakers, great, But it is pretty tough for full range speakers to really do it, so you are probably going to end up with subwoofers anyway, and remember as I said earlier, that in order to do subwoofers correctly you have to have full range speakers to begin with.

WSR Reber: I agree with you there. Do you subscribe to the Home THX specification for dissimilar speakers using direct radiators in the front hemisphere of the room and dipoles in the rear hemisphere of a multichannel system?

Vandersteen: No, unless you go to a THX movie theatre, and it is the most wonderful, most amazing, most incredible thing you have ever heard. The only justification for THX is if you have that opinion and you want to make a miniature version of it in your home. The best way to accomplish that is to go out and buy a THX-certified system. Most people with any high-end experience at all are able to go far beyond that in terms of realism and performance. There are people though, that really enjoy the theatre experience, and I think THX has done a lot for us in that regard, and has improved the sound in theaters dramatically. The industry is aware that THX is out there, and I think it has had an effect on the quality of the software that we get. So I am not anti-THX, I just think that it is not a high-end solution for the home. Many of the problems that THX tries to solve just don't exist in a home. These are theatre problems and I think that some of their choices were probably wise for the theatre. Some people want to duplicate the sound of the theatre in their home and some people take this to the point.of literally buying theatre chairs, popcorn machines and little booths. I mean if this is what you are trying to do then I think that THX is a very successful way to go about that, but if performance and accuracy in the home for two, four, five or six listeners and viewers at one time is your goal, then you would have to do nearly everything exactly the opposite of what THX recommends.

WSR Reber: Where should the surround speakers be positioned in the listening space? Optimally speaking now. Should these surround speakers be of the same type as the front speakers. Is a different model of speaker permissible for the center channel? We've touched on various things about this already, but now direct your comments directly to these subjects.

Vandersteen: I have best success with the surround speakers being, if not being identical to the front, at least having the same phase, time and amplitude response. What you need to remember is those three parameters determine what a speaker sounds like. So you should at least have the same phase, time and amplitude response for the rear channels so that these five acoustic fields can converge into one hologphonic sound space. Now the easiest way to accomplish that of course is to have five identical speakers because if there are phase idiosyncrasies in the design at least they interact with one another identically at all five corners. Now it is very difficult to design speakers that have the same phase, time and amplitude response in the real world, when used as they are going to be used. But I believe that is a very important criterion.

The center channel by its very definition needs to be designed differently than the right and left speaker because of the environment in which it will be used. However, its time, phase and amplitude performance should match the right and left speakers. By its very definition then, it needs to be significantly different in how it is designed from the right and left speakers in order for all three to perform the same in their different surroundings.

WSR Reber: Of course, unless you are going to build for yourself a five channel discrete music system using DTS Digital Surround, for example, limited to music CDs, and you didn't have a picture at all and you wanted to optimally reproduce multichannel discrete music then five absolute identical speakers would be best.

Vandersteen: Don't forget though, that we are saying that free standing speakers need to have anywhere from two to three milliseconds of delay before the sound strikes a sidewall. Remember, we said that the center channel needs to be further away from the listener than the left and right speakers, so that's automatically going to put that center channel closer to the front wall than the right and left so it still needs to be designed differently to have the same sound in its real world installation I want to keep coming back to that because in theory five identical speakers are what you need. What you really need is five speakers that sound identical to one another in three totally different environments. The rear channels are in one environment that 99 percent of the time is very different from the environment that the right and left loudspeakers are going to see, and then the third different design is going to be the center channel because of the way that it's normally used. Now, taking all that into account, the way that those three different things are going to be used, they need to sound the same even though they are used in totally different environments. Remember that the environment a speaker is in is 50 percent of how it is going to sound. You have to compensate for that or they will not sound the same.

WSR Reber: I agree, but if one was to build a dedicated room then the environment could be designed for optimum identical loudspeaker performance and speaker positioning.

This is a different question, but related. In a three way full frequency system the drivers have to converge at some point to be coherent. How do you design those speakers? At what listening distance from the actual speaker box do your drivers all coherently converge?

Vandersteen: Well, most of what we make are three ways, We group the drivers as closely together as possible so that the convergence point is as close to the speaker as possible, but you're still looking at six to seven feet before they are truly measuring right on an impulse, to where they truly are giving you an accurate facsimile of the musical waveform that was fed in. Now because our center channel and our on-wall speakers are coaxial designs, they accomplish that within a couple of inches after the sound leaves the drivers.

Coaxial loudspeakers have very, very severe design problems, especially if you are an anti-diffraction kind of guy. But there are applications where those disadvantages are outweighed by other factors. Sometimes the more important parameters of accurate phase and time response can't be achieved any other way.

"...the center channel needs to be further away from the listener than the left and right speakers..."

WSR Reber: To what extent is the digital 5.1 discrete channel format effecting your approach, both in terms of speaker design and application in room placement in the context of a multichannel, multispeaker system?

Vandersteen: It is not affecting me at all because I believe that accurate transducers need to be used even if they come up with a 7.1 system. So really it hasn't affected our design philosophy at all because I still believe the ultimate choice is always the highest quality transducer that you can possibly get and I don't see them being unique in any way because it's a 5.1 or Dolby ProLogic system, home theatre or music. An accurate transducer is an accurate transducer and it would be the ideal thing to use in any of these things that you are trying to accomplish.

WSR Reber: Do you have any other comments on the optimum approach to extracting the full potential of the 5.1 format, before we close out the interview?

Vandersteen: I think that a lot of the things that we have discussed will very definitely optimize the performance of 5.1. One of the comments I've heard is how, for music at least, Dolby Digital is not up to speed sonically. I will say this though, I have played around with it a lot now, and I find it to be a very good format for movies. In movies the fidelity is almost second to the effect, and the effect of discrete channeling, depending on what the engineer decided to record-the palette that he decided to paint with-is basically very, very good. You're visually distracted when you are watching a movie anyway. So I find 5.1 good enough to justify quality amplifiers and very high quality transducers in the system. However, if you were to turn the video off and start doing music only, that's a different story. I hope that we find a higher fidelity format for music, because I feel that Dolby Digital is inadequate.

WSR Reber: The 5.1 Dolby Digital is inadequate, that's what you are saying?

Vandersteen: Yes. You are saying the 5.1 format in general? I was mistaken. I thought in all cases when you were talking about 5.1, you were speaking specifically of Dolby Digital 5.1.

WSR Reber: No. The 5.1 format, I was speaking in terms of a 5.1 delivery system. Now what kind of fidelity we get out of that is depended upon...

Vandersteen: The system itself.

WSR Reber: Yes, exactly, whether it is Dolby Digital, DTS Digital Surround or whatever.

Vandersteen: I prefer DTS and I think that the potential there is wonderful. And I am also excited about other 5 channel formats, I guess my opinion would be: drop the "**.I**" but keep the 5. And once we have these five discrete high performance channels in our system, it's a simple matter in these expensive and very high quality processors to also take full advantage of the old matrix system. It would be so effective on all of our two-channel sources. And now you have a very high quality system that you can assemble over a period of time. If you do it in modules you could build a very, very high performance five-channel system that will basically take over two-channel music listening if it is done at a high enough quality level.

That is one reason why I advocate doing this a piece at a time. You know one of the problems with the insistence on 5.1 having five identical speakers is that by your very definition, if someone wants to go out and buy a system, you're driving the concept of mediocrity, because now instead of buying a good pair of speakers, people have to go out and buy five of them plus subwoofers. And I really believe that the modular building approach, where you build the system up correctly in phases over a period of time is the most successful long term way to go.

WSR Reber: I totally agree with you. The reason I put so much emphasis on an optimum approach is because I feel that the industry has failed the public in telling them what the optimum approach is so they can make reasonable decisions to establish a goal and then use the building block program to attain the end result defined by the goal, which for me is virtual acoustic holophonic reality.

Vandersteen: Absolutely.

WSR Reber: What has happened is the industry has started out with a compromise approach telling them that this is the ultimate when it is not. That is a real concern to me and why I have carefully reseached the 5.1 coding system alternatives and have determined that the DTS Digtial Surround system delivers the highest fidelity performance and is even more transparent than our current CD platform.

Vandersteen: As long as we realize that over a five or ten year period of time you want to accomplish some higher level of fidelity. I am glad that you are making this point and I agree with everything you are saying. We don't want to press this issue to the point where people will insist on buying five identical channels at one time.

WSR Reber: Yes, unless one can afford to do it right initially, optimally speaking.

Vandersteen: Yes, otherwise by its very definition it means that it is going to be lower quality than it would have been if they would have bought two high performance channels and added to it. Or took an existing two-channel system that was already very high performance and added to it a piece at a time.

WSR Reber: I agree with that.

Vandersteen: I think this is something that the industry hasn't looked at very carefully yet.

WSR Reber: Richard, this has been great. Thank you. ■

For information on Vandersteen Audio Loudspeaker Products and further advice please contact Richard Vandersteen at 116 West Fourth Street, Hartford, CA 93230 or phone 209 582 0324 or fax 209 582 0364.

This article has been reprinted in its entirety from Issue 24 of Widescreen Review, "The Essential Home Theatre Resource™". Subscriptions may be obtained at the rate of $25.00* for six (6) issues. Telephone our customer Relations Department at 909 676 4914 or toll free at 888 WSR SUBS for Visa, Mastercard, American Express or Discovery Card payment, or mail your check to: Widescreen Review, 27576 Commerce Center Drive, Suite 105, Temecula, CA 92590.

* Foreign subscription rates are $28.OOUS Canada and $50.OOUS International