stereophile

A REPRINT FROM APRIL 1996, VOL. 19 NO. 4

Linkwitz with the new Audio Artistry Beethoven

SIEGFRIED LINKWITZ

THE LOUDSPEAKER DESIGNER AND ALL-AROUND AUDIO ENGINEERING VISIONARY TALKS TO SHANNON DICKSON

Siegfried Linkwitz was born in Germany in 1935. He received his electrical engineering degree from Darmstadt Technical University prior to moving to California in 1961 to work for Hewlett-Packard. During his early years in the US, he did postgraduate work at Stanford University. For over 30 years Mr. Linkwitz has developed electronic test equipment ranging from signal generators, to network and spectrum analyzers, to microwave sweepers and instrumentation for evaluating electromagnetic compatibility. He has also had a long and distinguished second career as an audio engineering visionary. Along with Russ Riley he developed the famed, and widely used, Linkwitz-Riley crossover filter in the mid-1970s. Since then he has contributed several important technical papers covering a variety of measurement and speaker issues to such publications as the Journal of the Audio Engineering Society, Electronics (Wireless) World, *and* Speaker Builder.

Most recently, he has joined forces with fellow HP engineer Marshall Kay, CAD (computer-aided design) specialist Kurt Pasquale, and marketing consultant Tom Hoffman to form Audio Artistry. This three-year-old, North Carolina–based company is dedicated to developing and crafting speakers based on the accumulated insights and wisdom Mr. Linkwitz has gained over three decades of loudspeaker research. I spoke with Siegfried about some of these insights and experiences during the course of evaluating the Audio Artistry Dvorak, the review of which is found elsewhere in this issue. My first question concerned what had motivated Linkwitz to get involved in audio.

Siegfried Linkwitz: I grew up in a family in which music was very much appreciated. My father and brother played the piano, and although circumstances during World War II prevented me from learning an instrument, I've always had a love for music. After graduating from university and joining Hewlett-Packard to design electronics, it was only natural that I wanted to build audio equipment I could use at home, so I got very involved in building power amplifiers, FM tuners, preamps, and you name it—anything electronic I needed to reproduce music. Then I had the fortune of meeting some other engineers at HP who were similarly involved in audio, particularly Lyman Miller and Russ Riley. Lyman was very much into electronic design and making recordings while Russ built amplifiers and had a keen interest in speaker development. They really turned me on to investigating things even deeper, and loudspeakers, to us, were the most interesting and challenging area since so little was really understood about them. The speakers then on the market could certainly be improved, so we saw a real chance to make a genuine contribution.

Shannon Dickson: *Could you share with us some of the fundamental problems you and your colleagues encountered during the early attempts to improve speaker performance?*

Linkwitz: One of the problems at the time was that good test equipment wasn't available to us. Russ Riley developed his real-time ⅓-octave analyzer and a pink noise source which we used to make in-room measurements. I bought an early Advent speaker, measured it using the real-time analyzer, and consequently developed an equalizer to flatten-out its frequency response. That was a first attempt on my part. I then experienced a real surprise after we went to some local stores and heard the Electrostatic Sound System's ESS-7. It just sounded great, much better than the Advent. Naturally, I bought the speaker and took it home, but after measuring it, I was astonished—it measured very poorly! That led to a whole investigation into why it sounded so good but tested so badly.

We found out rather quickly how important driver quality was, as well as the distortion contributions of cabinet resonances. We began experimenting with wool stuffing in the box and with various bracing and panel damping techniques. We found that wool could be a very effective loading material. A number of commercial designs sounded much better when we replaced whatever they had inside with natural wool fiber. In my early designs, we tried two basic concepts built around rather small enclosures, both of which worked quite well. For instance, we made some very rigid, heavily braced small monitors; then we went the other way, using very limp, thin panels for the box construction. These were very easy to damp by applying roofing tar with sand mixed in. As you can imagine, this was a real messy operation—it smelled pretty bad too, particularly if you placed the speaker in the sun. It would out-gas for several weeks before you could tolerate the smell!

While it damped box resonances quite effectively, this approach was not really practical from a commercial point of view, nor would it have been a very welcome addition to most people's living rooms. But it did demonstrate how important minimizing box resonances is and just how difficult it is to really control this form of resonant behavior.

Dickson: *You've worked with some of the most respected engineers in audio over the years. Who had the greatest impact on your thinking regarding speaker development?*

Linkwitz: I mentioned Lyman and Russ already. Lyman was really into the recording side of things, so he did a lot of recordings on a semiprofessional basis and was particularly interested in capturing sounds as close to their natural origin as possible. So we had some great reference material to guide our evaluation. I learned a lot about recording from Lyman and continued to make many of my own reference recordings, which I used extensively during the development of these new speakers. Russ Riley is a very ingenious design engineer and, on top of it, a superb listener. I was always impressed by how easily he could identify just what the problems were in a speaker and in what frequency range and what one needed to do about them. He had absolutely superb hearing. While not as well-known as some of the other engineers, both Lyman and Russ had a big impact on my early audio career.

Through my work in developing test equipment for Hewlett-Packard, I met Laurie Fincham [then with KEF, now with Infinity] and we became good friends. We've shared a vast amount of information with each other over the years, have met frequently, and consequently had some very positive mutual influence on one another. Through Laurie, I was also introduced to a number of distinguished engineers such as Floyd Toole, Stan Lipshitz, John Vanderkooy, and Peter Walker from Quad. I had been following all of these people's writings very intensely all along, so it was a joy to meet them.

Most of these folks have been at my house at one time or another to listen to various ideas I had been working on. In addition, I have been an avid reader of the *JAES* throughout the years, as well as *Wireless World* from the UK [*now* Electronics World—*Ed.*]. *Wireless World* used to carry a great deal of high-quality information about audio and speakers; it still does, in fact, though it's not as easy to find these days. Actually, my first publication appeared in 1978 as a lengthy three-part article in *Wireless World* in which I described the construction of a three-way active speaker system consisting of small satellites and a subwoofer.

In summary, the various influences on my thinking have led to a general approach that is really a blend of the analytical—meaning the measurement of things—and the subjective listening experience, to try to find out what is really going on. If there is a hypothesis of why something works—this way or that—I'll set up an experiment to see if I can prove it or disprove it. In this way, I've always attempted to correlate what we hear with objective measurements—not always successfully, mind you, but at least making the connection where possible. This method will give you a lot of insight into which measurements or artifacts are important and which are not so important. Occasionally, I've found results that look very significant on paper but are barely perceptible, if at all, while on the other hand, some extremely slight irregularities can be very important sonically.

Dickson: *Can you tell us what your priorities are in making and evaluating specific measurements?*

Linkwitz: I've learned there is a whole battery of measurements one needs to use—and interpret correctly—in order to get a better picture of any given speaker. No one measurement will tell you the whole story. At the top of the list is definitely a loudspeaker's on-axis anechoic frequency-response measurement because this represents the direct sound you hear. However, of similar importance are the vertical and horizontal anechoic off-axis responses. So in my designs, I try to achieve a very well-behaved off-axis response which duplicates the shape of that on-axis, but steadily decreases in level the farther you move off-axis. This is so important in determining the reverberant field and the reflected sound in the listening room.

Another key factor I learned during the development of my crossover design is that, when two drivers are combined in the crossover region, their summed output should be at its maximum on-axis. In other words, the radiation pattern remains stable at the crossover region and doesn't shift. For example, I've found through experimentation that it is definitely audible if you go some distance above-axis and all of a sudden have a maximum peak or sharp dip in the crossover region. This problem is similar to what happens with many large-panel dipole designs. As they produce higher frequencies, their off-axis response becomes more irregular, with peaks and valleys that can color the overall sound and make speaker placement in a given room very critical. If the crossover on any speaker doesn't blend together, you can get this kind of off-axis peak.

Another measurement I look at is the overall frequency response on a half-octave or octave basis, just to see the general trend: whether the treble is rising or sloping, etc. When you look at any response in detail, you never get a flat picture, you always have little ups and downs; but I've found you don't really gain anything by trying to smooth out these small ripple effects in the response. However, how smooth the response is over a third- or half-octave basis *is* important. I'm essentially looking for an averaged-flat anechoic response.

I do my quasi-anechoic measurements outdoors, with the speakers mounted on a 50" turntable so that the speaker is as far away from any reflecting surfaces as possible, yet still manageable. I try to get 10 milliseconds of undisturbed sound between the initial impulse response and the arrival of the first reflection, which will give me a frequency resolution of 100Hz and useful data for all frequencies above a couple of hundred Hertz. I also try to minimize the first reflection off the floor or ground with acoustic absorbers.[1] But as you can see, this method really doesn't tell you much about the bass.

After my series of anechoic tests, I perform in-room measurements over a 50ms time window. This gives me a frequency resolution of 20Hz, and since 50ms is a pretty long time in a room, it does takes into account the room reflections. I also use 50ms because that is about the maximum time span [during which] the human brain can process the characteristics of a sonic event. Basically, I use these in-room measurements as confirmation of the anechoic results, not to correct for all the reflection anomalies or peaks and dips that show up in the response. I do, however, make these in-room tests from several different locations, and with our new dipole designs, even these in-room measurements over a long time window are surprisingly smooth and flat.

WE FOUND OUT RATHER QUICKLY HOW IMPORTANT DRIVER QUALITY WAS.

Another test I perform looks for resonances and stored energy in various locations—using a Shaped Tone Burst stimulus, which is particularly well-suited for this. This is a *tremendous* test signal. I measure the impedance curve of the drivers themselves to reveal driver anomalies, and I also use complex multi-tone signals to test for nonlinear intermodulation distortion artifacts.

Dickson: *This is essentially the spectral contamination distortion measurement you are speaking of?*

Linkwitz: Yes, exactly, the same concept, in order to find nonlinear problems. Interestingly, in the old days when we used pink noise as the stimulus to try to equalize a speaker to be flat at the listening position in a real room, it typically turned out too bright-sounding. This is an approach that may be useful in a PA setup, but in a listening room it doesn't lead to a correct result.[2]

Dickson: *Tell us more about the Shaped Tone Burst test you just referred to. I found your article in the April 1980 issue of the* JAES *(Vol.28 No.2), discussing the benefits of using this stimulus in speaker evaluation, very interesting.*

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1 Color me envious. An anechoic time window of 10ms is excellent. Performing speaker measurements in the *Stereophile* listening room, with its 9' ceiling height and a microphone distance of 50", results in about a 3ms–4ms anechoic time window, this with a very thick pile of absorbing material on the floor. Siegfried must live in a very quiet neighborhood; there is too much background noise in Santa Fe to perform measurements outdoors and get usable waterfall plots. —JA

2 This is because you are equalizing the loudspeaker's power response, which includes the full contribution in-room of its off-axis behavior. As the power response tends to slope down with increasing frequency with conventional speakers, such equalization will boost the highs on-axis. As a result, unless you are sitting a *very* long way away from the speakers, the perceived balance will have a strong contribution from the speaker's direct sound which, after equalization, will tend to be too bright. —JA

Linkwitz: From a practical standpoint, the advantage of using a shaped tone burst (one that rises and decays gradually in a sinusoidal envelope) is that all of the burst energy is concentrated into a very narrow frequency band. This is quite different from tone bursts used in the past, where you had a rectangular burst covering a fairly wide frequency band. I chose a spectrum width of a third of an octave for this stimulus — which is a 5-cycle burst — because this corresponds closely to how we hear. A third-octave is about the width of the critical band of hearing. Also, because the burst is so short in duration, you mask out the effect of reflections, so it becomes a sort of poor man's approach to anechoic measurements. As long as you measure the peak of the burst before the first reflection, you've essentially captured an anechoic-like response giving you some of the benefits of Time Delay Spectrometry or Maximum Length Sequence (MLSSA) techniques without the expense.

Now, the shaped tone burst can be used in several ways. For instance, one can just use a microphone to measure the peak amplitude that the burst reaches after you apply it to a speaker, which will give you an approximation of the frequency response. Likewise, after the decay of the 5-cycle burst, there shouldn't be any output from the speaker. In reality, however, if there is stored energy in the drivers or cabinet, the speaker keeps on ringing. Therefore, the shaped tone burst is very useful for identifying the sources of resonant storage. In any event, I do get extremely good correlation between the frequency-response measurements derived from the shaped tone-burst test and what we hear, as well as specific information about cabinet and driver resonances.

The real benefit of this type of test is that it concentrates the energy into a constant narrow frequency band so that it is a third-octave in width at 100Hz or 1kHz or 10kHz. Therefore, it is much narrower on an absolute basis at 100Hz than at, say, 10kHz. In other words, the tone-burst test has a constant resolution on an octave basis. This is important when you compare it to FFT analysis, where you get good resolution at high frequencies but very little information at low frequencies. The shaped tone- burst test works on a logarithmic scale so we can get good resolution all the way down to the lowest frequencies. I use this type of test signal to look at the decay of the burst, which gives me the same type of information that you would be looking for in a spectral-decay or waterfall plot that MLSSA can generate.

I also have MLSSA, so I do generate the spectral-decay plots as well, but I have to say, I have not found the waterfall plots very useful except for maybe above 1kHz. Below 1kHz there are so many artifacts in the typical spectral-decay waterfall plot that it is useless. Anyway, it's simply a lot easier to get the same, and even much more, information out of the shaped tone-burst response. Extending the time record for the FFT in order to get useful low-frequency data is generally not practical; using a narrow burst signal makes it so direct and easy. Plus, you can change the frequency of the tone burst on the fly, while you watch the dynamic changes on an oscilloscope, as the tail of the burst stretches out — in effect allowing you to see directly when you're close to a resonance!

I guess I'm beginning to sound a little like a missionary for the shaped tone-burst test, but I really do believe it is an extremely powerful technique that is too infrequently employed. Many people are just not aware of how it differs from traditional tone-burst stimuli. Today it is particularly easy to generate the required burst signals since you can buy an arbitrary waveform generator fairly inexpensively. Also, it would be very easy to include a series of 5-cycle–wide bursts at various frequencies on a test CD; then, with an oscilloscope or perhaps one of the PC-based software test systems, the audiophile would be equipped with a powerful tool for evaluating his system and speakers.

One final attribute of the shaped tone burst that I find very important is that it's a particularly safe signal with which to test the maximum output of components. For instance, if you use a burst rate of 1Hz with a 5-cycle burst you'll have a very low duty-cycle, so even if you require 100 watts to clip your tweeter, the short duration of the burst — it's essentially like a frequency specific pulse — will prevent you from overheating the voice-coil and damaging the driver.

Dickson: *You're most widely known as the developer of the Linkwitz-Riley crossover. Could you explain a few of the characteristics of this crossover?*

Linkwitz: To answer your question, we need to go back to when I started out exploring the whole speaker issue in the early '70s. Then you could take the grillecloth off many of the available speakers and see a strange, almost haphazard arrangement of the drivers on the baffle. It really puzzled me and I wondered what was going on. So I asked some of the designers why they were doing this and they said, "Because we've found it sounds better."

As I looked further into this issue, I realized that two principal things were not well-understood. First, very little was known at that time about the effects of diffraction from the cabinet edges. Second, and more importantly, very little was understood about how phase-shift with respect to the current passing through the voice-coils of different drivers affected the polar radiation pattern of a speaker. In other words, the interaction between the electrical side of a driver and the acoustical response was not clear at the time. For example, the phase-shift between the current in the tweeter and midrange voice-coils, relative to the placement of these drivers on the baffle, affects the speaker's radiation pattern.

Basically, since few drivers are really coaxial, with the difference in physical placement — that is, if the path lengths between the drivers and the listening point are different, or even if they are the same — you get a vector addition which is a function of the phase-shift between the different voice-coil currents and the distance between each driver and the listener. So Russ Riley and I began our work, in earnest, to be sure that the drivers were in-phase in the crossover region. This, in essence, is what the Linkwitz-Riley crossover is all about: making sure that you have the same acoustic phase between the midrange/woofer and the tweeter at the crossover.

Dickson: *How about the phase relationship outside of the crossover region?*

I'VE ALWAYS ATTEMPTED TO CORRELATE WHAT WE HEAR WITH OBJECTIVE MEASUREMENTS.

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Linkwitz: As it turns out, that same phase relationship is maintained at other frequencies as well. This is very much in contrast to the classical Butterworth crossovers that people use in a number of speakers. An inherent property of the Butterworth design, whether these are first-order, third-order, fifth-order, etc., is that the crossovers are always in phase quadrature. In other words, the acoustical signals coming from the midrange and tweeter are phase-shifted by 90° relative to each other. At its –3dB point, each driver has an amplitude of 0.7, and if you add two 90° phase-shifted vectors of 0.7, you get unity — the outputs of the two drivers add to unity on-axis. However, as you move farther away off-axis, one or the other driver will experience more phase-shift as the path-length difference becomes longer, and you'll have either a dip or a peak in the amplitude response off-axis.

In any event, the true maximum output of the two drivers will occur someplace off-axis, and this is an audibly bad thing. The peak off-axis response can then reflect from the nearest boundary and combine with the direct sound as added coloration.

Now, a first-order crossover can be made phase-perfect at one point in space, but I feel quite strongly that you cannot just look at a speaker's performance at one single point in space. The off-axis response is also very important to a speaker's overall performance in a real room, because the radiation in these other directions will add, through reflected and reverberant interactions, to what you hear. Typically, we don't listen to speakers outdoors or in anechoic chambers.

For an ideal Linkwitz-Riley crossover, the amplitude is flat on-axis or at unity, just as it would be for an ideal Butterworth. However, the Butterworth response will have its peak off-axis. In contrast, the amplitude of the L-R crossover will be down in level off-axis, and will never be higher than the on-axis response. The crossover point of a Linkwitz-Riley will also be at the –6dB point, equivalent to an amplitude of 0.5, and only when you add vectors with amplitudes of 0.5 that are in-phase will you get unity. If there is any phase angle between these half-amplitude vectors, their sum will be less than unity.

A very important point that people sometimes miss in this discussion is that when we are speaking of a given crossover, we are talking about an *acoustic* crossover, or what happens acoustically. Now, what I have to do electrically to achieve the correct acoustic response may not look anything at all like a textbook filter design. The actual filter often looks very little like the drawings I may show to explain any given example. This is also true for a Butterworth filter. It is highly unlikely that a textbook electrical Butterworth crossover will produce an acoustic Butterworth response, because the driver's response enters into the picture as well.

Dickson: *There is a general misconception in some circles about differential* vs *absolute phase effects in speakers. Recently, I've heard about some well-meaning but misinformed retailers who arbitrarily reverse the polarity of either the tweeter or midrange hookup wires in all of the speakers they sell that are designed with high-order crossovers, in an attempt to make them "in-phase" — much to the horror of the original designer. Perhaps you could shed some light on this issue.*

> THE SHAPED TONE-BURST TEST IS AN EXTREMELY POWERFUL TECHNIQUE THAT IS TOO INFREQUENTLY EMPLOYED.

Linkwitz: If someone were to arbitrarily change the polarity between drivers in a good Linkwitz-Riley crossover, they should get a strong null at the crossover point on-axis. In fact, this is a test I use to see how well I have executed the acoustic crossover. However, making such a change with the idea of somehow making a "phase-coherent" speaker is not correct. It will certainly change the sound, mind you, but is definitely not recommended.

Dickson: *The Dvorak and Vivaldi speakers represent a radical departure from your earlier philosophy. What inspired this change in direction, and could you outline some of the primary goals you've tried to achieve with these new dynamic dipole designs?*

Linkwitz: I would have to say the departure in my thinking happened by coincidence. At the time, I had volunteered to build a public address system to improve speech and sound intelligibility for a video production in a large, highly reverberant gymnasium. I designed a long directional column speaker with multiple 6" dynamic drivers firing as dipoles. In other words, the back of the column's baffle was open so the sound radiated to the front and rear with each direction out of phase with the other. The directivity of the radiation pattern of this design worked really well in this very reverberant environment. You could understand what was said just as well from the back of the hall as from the front. Well, just for kicks, I took the thing home, split this long column into two shorter stereo columns, and decided to see how it sounded in my living room. I was really surprised to hear that it had some of the qualities that impressed me so much with other dipole speakers, like the Quad electrostatics. Not that these PA speakers were anywhere close in tonal balance or transparency, but there was something fundamental about the character and quality that reminded me of these better dipoles.

In any event, that got me started on investigating the possibilities of using conventional high-quality dynamic drivers as dipoles. Quite frankly, I've always been very fond of certain characteristics that some electrostatic dipoles possess, yet I had never seriously pursued panel dipoles, in spite of their good qualities, because there was not enough dynamic low-frequency output for me. They also had a very tight listening spot, and were generally just too limited in what they could do.

Based on my initial experience with the PA columns, I set out to build a speaker that could perform similar to an electrostatic dipole but using conventional dynamic drivers in a manner that would avoid some of the limitations faced by large panel designs. It took several evolutionary iterations of the design to get a good understanding of which aspects of dipole operation are really important and which are not so critical, and why this is so.

During this development phase, I discovered how important the baffle shape and design were, as well as the frequency range over which dipole radiation was most beneficial. For instance, I found that dipole radiation of the treble region was not only unnecessary but, in fact, a disadvantage. Interestingly, you don't even see the rear-firing response from a dipole tweeter when you measure it on-axis, but when I listened to it in the room, I found that it caused some high-frequency "splatter" that didn't seem natural. I abandoned that approach and used a monopole dome tweeter from 2kHz on up and dipole radiation from the cone drivers down to 20Hz. I started out with a completely active system, with separate amps for all drivers, and used equalization as well. I equalized the speaker to be flat since a dipole on a small baffle has this natural 6dB/octave rolloff below a certain point. The first version was a pretty elaborate prototype.

Actually, at first I used a closed-box woofer because I didn't think I could get enough bass output from a compact dipole woofer system with a reasonable number of drivers. This is because of the acoustic short-circuit between front and back that a dipole represents. However, after sharing notes with Brian Elliott, a good friend and acoustician who had developed a splendid dipole bass system using six 12" drivers per channel, I began looking at the possibilities for dipole bass more closely. His system simply produced the most astoundingly natural low-frequency reproduction I had ever heard. However, I still stayed with the closed-box woofer concept a while longer. Then Don Barringer, another good friend and the recording engineer for the US Marine Band, reported great results in a normal-sized listening room using a speaker based on my design, but extending the dipole operation into the bass single stereo amplifier.

Dickson: *I suppose the directivity of the Dvoraks, particularly with the separate dipole subwoofers, presents some special considerations when these speakers are measured.*

Linkwitz: One of the real problems was: How do you measure a dynamic dipole accurately? For instance, you cannot measure close to the cone because then you'll only see what the cone does essentially. You will not get the effect of the cancellation when the negative-polarity rear wave combines with the forward positive wave, so you have to measure some distance away. That really forces you into either an outdoor or anechoic-type measurement. It's very important to do it this way to get meaningful results. If the cabinet is large, or if the drivers are spaced very far apart, you have to have at least the same distance to your measurement microphone in order to capture the integrated sound coming from the rear, or off the edges of the cabinet, etc. The separate subwoofers are quite tricky to measure as well.

The bottom line on the Dvorak is that everything is based on a flat response under anechoic conditions with a moderately directional radiation throughout its full range. When I say that the dipole aspects of the speaker are directional, it's not in a very strong sense. For instance, the response is 1dB down at 30° off-axis, 3dB down 45° off-axis, 6dB down at 60°, and has a null around 90°. The monopole tweeter also maintains a similar directional characteristic because of the baffle

IT IS HIGHLY UNLIKELY THAT A TEXTBOOK ELECTRICAL BUTTERWORTH CROSSOVER WILL PRODUCE AN ACOUSTIC BUTTERWORTH RESPONSE.

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with two 12" drivers per channel equalized flat.

So, while I was refining the dipole midrange/monopole tweeter parameters of my speakers, I was actually a latecomer in taking the dipole concept all the way. As it turns out, and somewhat surprisingly to me, two big dipole woofers per channel does a very respectable job. Not quite like Brian's system with its total of twelve 12" drivers, but actually very good. On the other hand, we've also learned that there is very little difference between four and six 12" woofers per side, and with some creative mounting techniques, we've been able to mount four large drivers in a surprisingly compact dipole enclosure for use in larger rooms, and as a standard feature in our new top-of-the-line speaker system, the Beethoven.

Anyway, the production Dvorak is a five-piece, actively bi-amplified system comprising two main panels, two separate subwoofer cabinets, and an active crossover/equalizer. It covers the full audible range and beyond, from 20Hz to 25kHz. If a person has a small room or doesn't require deep bass below 40Hz, they can just use the main panel with a single amplifier, but the active crossover/equalizer is still required for the EQ of the two mid/bass drivers. We have also recently released the Vivaldi speaker, which has the same driver complement as the full Dvorak system but all mounted in one tall speaker. The system uses a passive design for both the crossover and equalization so it has a reasonably flat response to about 40Hz, below which the dipole's natural rolloff occurs. This system is ideal for those in small- to moderate-sized rooms who don't need the full 20Hz extension of the complete Dvorak system. It also requires just a design and the wavelengths of its frequency band. Of course, it differs from the dipole drivers in that it fires predominantly forward. The important thing is that the shape of the off-axis room response is very consistent with that produced on-axis, resulting in the open-sounding soundstage and the speaker's even tonal characteristics. Also the dipole "figure-8" cosine directionality goes all the way from 2kHz down through the woofer's range to 20Hz. This directional deep bass is really pretty amazing! If you have another person to help you, play some low-frequency tones, then have your friend rotate the woofer cabinet and you can clearly hear the output null at 90° off-axis!

Dickson: *Yes, I've noticed I can hear increased focus in the low bass when I toe-in the woofers. This is one speaker where "stereo" bass may have some real meaning. Two of the Dvorak's sonic characteristics I find most striking are its dramatic reduction of room-induced colorations from the low bass through the midrange, and its ability to convey image height in correct proportion to the width and depth dimension of the soundfield. What factors do you think contribute most to these effects?*

Linkwitz: These are primarily due to the dipole characteristics and the even room response. Since the speaker is moderately directional at all frequencies, more of the sound is directed toward the listener and less to the walls and ceiling. Therefore, less comes back from the room in the form of resonances or reverberation which will blend with and color the direct sound from the speaker.... The active equalization is merely there to correct for the dipole cancellation that the raw drivers would have if you didn't compensate for the inherent 6dB/octave rolloff.

Now the image height is an interesting thing. I have to state that I don't fully understand all the psychoacoustics involved here, but I have found that it is important that the center of the radiating elements be at about ear height, and that the speakers have some vertical extension as well. I have built many small two-way minimonitors; while these systems can have very nice horizontal dispersion and excellent imaging, I've always felt that I was listening through a horizontal window, one that was very wide but with a height not much greater than that of the speaker itself. It's like listening through a horizontal sliver. Now vertically spreading out the driver's arrangement expands the vertical dimension of the soundstage and adds much more realism for me.

With respect to the reduction in overall room colorations that the Dvoraks provide, that has a lot to do with how the dipole characteristics are implemented. This comes back to the fact that the off-axis response is very well-behaved in this system. In other words, the design concentrates just as much on the off-axis performance as on the on-axis. While I don't have any definitive proof, I strongly suspect that the erratic off-axis behavior of most panel speakers is what makes their room placement so critical, forcing a person to locate the panels in a place that minimizes reflections and changes how the off-axis sound couples with the room, in order to get a balanced output. On the other hand, quite frankly, I have not found the performance of either the Dvorak or the Vivaldi to be critically dependent on room placement compared to other speakers. There is still definitely an optimal placement in any given room, but you can get very satisfactory performance in a wide variety of locations, so this experience lends further credence to the value of a well-behaved off-axis response.

Dickson: *While the Dvorak and Vivaldi represent a somewhat fresh approach to speaker design, they are mature designs. I'm very curious to hear about what projects you have planned for the near and more distant future.*

Linkwitz: Recently, I've been doing extensive investigations into numerous drivers using some of the newly developed measurement techniques I alluded to earlier—especially the tests for nonlinear distortion artifacts, and those that help locate and define energy-storage effects in drivers themselves. All this in a search for components that have even more clarity and transparency. What I had in mind was to see how much further this Dvorak concept could be refined.

We unveiled our new flagship, the Beethoven, at the recently completed '96 WCES in Las Vegas. In addition to an all-new balanced electronic crossover, each main panel has a new silk-dome tweeter, two new 8" drivers, and a pair of 10" dipole drivers—all low-distortion, high-excursion models. Both of the woofer cabinets for the new system contain four 12" dipole drivers, so obviously this system is designed for high-output, very-low-distortion sound and will be considerably more expensive than the standard Dvorak. I must say we have been extremely gratified with the performance of the new system. So that's one project we are putting the finishing touches on now, and we are also thinking about a smaller, lower-cost version of a dipole speaker in the future.

A little farther down the road, possibly over the next few years, I would also like to settle in my mind the importance of what I would describe as "linear phase." This refers to obtaining a result that is a more accurate replica of the time-domain wavefront. Some people seem to think that this is very important for reproducing clicks and transient-type sounds, and that may well be. From a common-sense point of view, it seems logical that you would want to have a true replication of the wavefront.

FROM A COMMON-SENSE POINT OF VIEW, IT SEEMS LOGICAL THAT YOU WOULD WANT TO HAVE A TRUE REPLICATION OF THE WAVEFRONT.

However, I'm not totally convinced because I have done a lot of experiments with phase-distorted signals. Basically, I've shifted the phase between different spectral components by running various signals through an "all-pass" filter, where the amplitude is unaffected when I change the phase response with frequency. When you look at these signals on an oscilloscope and change the phase, they look grossly different, so you'd think "surely this *must* sound different." But when you listen, you can't hear the difference, even though the time-domain waveform staring you in the eye looks so totally altered!

I did quite an investigation into this when I initially developed the Linkwitz-Riley crossover because it is not a linear-phase system—nor are the Butterworth crossovers, for that matter, except for the first-order slope. The experiments I've done so far have not convinced me that phase distortion in small amounts is audible. Now if the phase distortion is gross, you can definitely hear it, but the typical crossover is far from producing that much phase distortion. However, some people whom I respect seem to think this is something that could have audible consequences, so I'm keeping an open mind about it and want to determine once and for all its value, if I can. I must say that I have not heard an example of a speaker design that conclusively demonstrates the benefit of a linear-phase system.

Dickson: *I imagine when you look at the total performance of a speaker with the various tradeoffs required to achieve a certain goal, you have to weigh their relative merits.*

Linkwitz: This is true. You could question, for example, whether the extra stress on drivers and resulting distortions produced by a first-order system are not more audibly significant than the subtle improvements potentially created by its linear phase effects. However, it is possible, using digital techniques, to correct for the phase response as well, and my friend, Malcolm Omar Hawksford [of England's Essex University], who has done quite a bit of work in this area, has kindly offered to perform a phase correction for the first 10ms time record of the Dvorak's impulse response with his digital processor. It certainly would require a bit of horsepower to implement digital phase correction in the active crossover, but it could be done.

Again, it's not yet completely clear whether a digital crossover will buy you anything. It *may* buy something, and that's the part I'm interested in. For instance, with this scenario we could combine the excellent on- and off-axis amplitude response of the existing Linkwitz-Riley crossover in the Dvorak with an after-the-fact digital correction of the time-domain, to achieve a linear-phase system. You see, the digital time-domain correction would not affect the existing passive or active crossover response at all, it just would correct overall phase. As a matter of fact, at a recent AES convention, both Malcolm Hawksford and I attended a discussion about the use of very steep crossover filters with digital phase correction. Convincing arguments were presented showing that these extremely steep filters produce sonic anomalies, and consequently are not desirable. Malcolm also stated that something like the Linkwitz-Riley fourth-order crossover was about optimum, even digitally implemented, when phase correction is applied. Anyway, I'm very interested to see how this research turns out.

Dickson: *With the continual improvement in driver technology and refinements in other areas of audio design, it may be that these more subtle issues, like linear phase, will become more important in the future.*

Linkwitz: I think that's a good way to look at it. You could say that you need to have a certain number of other things done correctly first before those effects come into play. I should also point out that the digital phase compensation I'm speaking of is very different from the digital room-correction systems you may have read about. In any event, these are a few of the areas that we at Audio Artistry look forward to investigating and developing in the near future. **S**

EQUIPMENT REPORT

AUDIO ARTISTRY DVORAK LOUDSPEAKER SYSTEM

Shannon Dickson

Main Panels: Drive-units: two 8"-cone dipole midrange/bass drive-units, 1" aluminum-dome tweeter. Frequency range: 100Hz–25kHz, extends to 40Hz with subwoofers turned off. Minimum amplifier requirements: 35Wpc tube, 100Wpc solid-state. Nominal impedance: 8 ohms, 3.2 ohms minimum at 2.1kHz. Dimensions: 56" H by 12" W by 4" D, mounted on 2.25" base. Weight: 50 lbs each.

Subwoofers: Drive-units: two long-throw 12"-cone dipole drive-units each. Frequency range: 20Hz–100Hz, with crossover. Minimum amplifier power: 25Wpc. Nominal impedance; 6 ohms, 5.5 ohms minimum. Dimensions: 27.5" H by 11" W by 20" D. Weight: 55 lbs each.

Crossover/Equalizer: unity-gain, non-inverting. Subwoofer level: adjustable to –12dB, plus additional –12dB with internal switch. Dimensions: 17" W by 8" D by 1.75" H. Weight: 10 lbs. Power supply: 5W "wall wart."

Overall frequency response: 20Hz–25kHz ±2.5dB. Overall sensitivity: 88dB/W/m (2.83V). Radiation pattern: dipole below 1.5kHz. Recommended room size: 250–600ft². Prices: total system: $5990. Main panels and active crossover without subwoofers: $3995. Subwoofers with unequalized crossover for use with other speakers: $3590/pair. Approximate number of dealers: 12. Manufacturer: Audio Artistry, 8312 Salem Drive, Apex, NC 27502. Tel: (919) 319-1375. Fax (919) 319-1416.

I first heard Audio Artistry's Dvorak loudspeaker during the 1994 WCES. I was leaning against the wall in a corridor of the Sahara Hotel's bi-level complex, trying to avoid being run over by swarming hordes while shootin' the breeze with Corey Greenberg. I was thankful for the respite from what seemed to be an endless succession of rooms playing Eric Clapton's "Tears from Heaven." As happens at most CE Shows, a particular tune emerges as the "official" demo track and gets saturation coverage. "Tears" was definitely the one for the '94 Hi-Fi lovefest. It's a great song—but after hearing it 30 times in a row, you do start to feel a little tight around the collar.

Chatting with Corey, on the other hand, is *always* fun, so when a nearby door opened up, filling the hallway with the very same song, you'd have thought I'd hardly notice. On the contrary, I immediately lost my train of thought, said "Aloha" to Corey, and drifted into the room like a Stepford wife heeding a subliminal message. By the last refrain of Clapton's poignant ballad, I felt like I was hearing the song for the first time. During the next hour's demo of classical, jazz, and rock tunes, I was genuinely enthralled by the way these speakers conveyed the essence of whatever music they were reproducing.

The impression of that first experience was so strong and persistent that I just had to see for myself if this speaker really *did* possess a unique and compelling communicative skill.

ARTISTRY

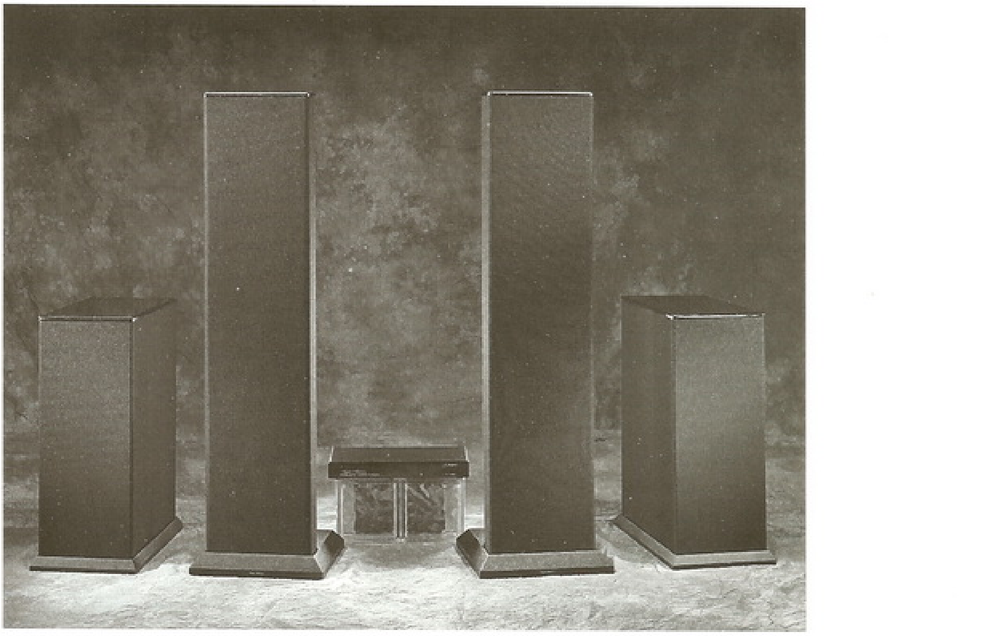
Audio Artistry is located in a suburb of Raleigh, NC and was formed almost three years ago by president Marshall Kay, a test and measurement applications engineer for Hewlett-Packard specializing in the application of HP's wide variety of electronic test equipment. Marshall teamed up with Kurt Pasquale (an expert in computer-aided design) and Tom Hoffman (owner of a local high-end retail store) to build, refine, and market a line of speakers designed by co-partner Siegfried Linkwitz.[1] Mr. Link-

1 The Audio Artistry Mozart loudspeaker was reviewed in the August '94 issue of *Stereophile*, but it is not a Linkwitz design. —JA

witz is a senior design engineer for Hewlett-Packard [*see the interview elsewhere in this reprint*], and is the same Linkwitz of the widely used Linkwitz-Riley crossover topology.

The challenge

I'll paraphrase the introductory abstract found in Linkwitz's 1992 AES paper, in which he describes the prototype for the Dvorak in great detail: "A relatively small-sized dipole loudspeaker system has been developed using conventional cone-type drivers to obtain sufficiently large volume displacements. The 3-way system has *dipole directional characteristics over the 20Hz to 1.7kHz frequency range for reduced interaction with the listening room.* Effects of baffle shape upon the radiation pattern have been investigated. Active crossovers and dipole specific equalization have been used to obtain a flat frequency response." (My emphasis.)[2] This rather curt, matter-of-fact description contains a wealth of information about the Dvorak's unusual characteristics.



Audio Artistry Dvorak loudspeaker system

The inspiration for the Dvorak arose from two principal factors. First, about nine years ago, the designer's reference speaker consisted of two small satellites and a central subwoofer. This system was capable of good dynamics, image specificity, and fine detail, as well as an even midrange/treble tonal balance and decent reproduction of instrumental timbres. But the design suffered from limited image height and a soundstage width restricted to the area between the two satellites. Additionally, no matter how small and dead the enclosure was made, some boxy coloration was inevitable due to resonant energy storage and its subsequent delayed release. Most important, all traditional box speakers radiate low frequencies in an omnidirectional pattern, something that Linkwitz felt produced unacceptable masking and colorations.

Faced with these shortcomings, Siegfried received his second impetus for the Dvorak design in 1986, when he designed a sound-reinforcement speaker for audio-video presentation in a large, highly reverberant gymnasium. In order to improve speech and music intelligibility in this environment, he built a long, vertical line-source dipole consisting of twelve 6" cone drivers with the drive-signals electrically tapered to concentrate high frequencies in the two center drivers of the array. In spite of concrete walls and a large parquet wooden floor, this system produced excellent clarity and speech recognition even in the rear of the gym.

As an experiment, Siegfried then broke these dipole columns into a stereo pair and set them up in his living room. Despite the speaker's skewed frequency response, the sound was surprisingly open and produced outstanding image height and width, as well as a marked reduction in room interactions through the midrange and bass. These findings inspired a major revision in his thinking—away from the semi-point-source monopole designs popular for home playback and toward a moving-coil speaker that would maintain dipole directivity from the midrange through the lowest frequencies. Numerous conversations with acoustician Dr. Brian Elliott, and subsequent work by Elliott on the challenge of building an effective dynamic dipole subwoofer, dovetailed nicely with Siegfried's efforts from the midband upward. The full-range Dvorak began to take shape.

Dipoles & monopoles

In order to better understand the Dvorak, we must examine the basic relationships between panel dipoles, moving-coil box speakers, and the Dvorak's successful combination of the two principles—with a particular emphasis on room/speaker interactions. (Note: The attributes and limitations detailed below are relative comparisons between the various major speaker types I'm familiar with, rather than judgments rendered against an absolute yardstick.)

We are still a long way from the perfect transducer. All sorts of complex intermodulation and harmonic distortions still exist which—along with limitations in driver technology, diffraction problems, crossover-related dispersion anomalies, and speaker-room interactions—plague even our most advanced designs. Not to mention the drawbacks of two-channel stereo. Please keep this in mind when interpreting the following comments.

Although a thorough examination of room acoustics is beyond the scope of this article, I'll be referring often to three fundamental interactions that each play a large role in what we actually hear from every speaker design: 1) Room resonances typically affect the region below approximately 200Hz and result from standing-wave excitation of room modes. 2) Initial reflections from room boundaries often negatively affect imaging and the resolution of transient detail. 3) Reverberation refers to longer-duration sounds resulting from cumulative reflections, and affects timbre principally above 200Hz. Let's look first at how these phenomena impact dipoles.

Dipole Basics: A dipole can be modeled as two independent point-source transducers separated by a gap measured between the front and rear center of the transducer, including the width and depth of any baffle. The two sources radiate opposite-polarity but equal-amplitude soundwaves, giving rise to a moderately directional "figure-8" dispersion pattern.

Some of the earliest known speakers were dipoles; by the mid-'80s several companies were producing large-panel dipole designs, many of which were, and still are, considered among the best trans-

2 Siegfried Linkwitz, "Development of a Compact Dipole Loudspeaker," presented at the 93rd Audio Engineering Society Convention, San Francisco, 1992, preprint 3431. Available from the Audio Engineering Society: Tel: (212) 661-2355. This paper provides a wealth of detail about the development of the Dvorak's prototype. Note that production versions of the speaker contain several important refinements, particularly in the crossover slopes and EQ employed.

ducers in the world. The midrange performance of the Quad ESL-63, for example, closely matched Siegfried's expectations in terms of transient response, timbral purity, and transparency.

However, using a large panel to implement a dipole radiation pattern is fraught with limitations: curtailed dynamic impact, the difficulty in achieving realistic bass reproduction, and the critical nature of speaker-placement requirements in order to achieve accept-

Dipoles

The following "visual aids" illustrate how the Dvorak's radiation pattern differs from that of other major speaker types. Fig.1 shows the polar response patterns of the three major speaker types plus that of the Dvorak/Vivaldi speakers. The diagram is divided into the three main frequency bands to illustrate the consistency, or lack thereof, in the dispersion pattern for each speaker over the full audible band. Note that the shaded areas represent approximate sound pressure level (spl) distribution in the horizontal plane, and don't reflect changes in dispersion behavior around the transition region between the three frequency bands. Also, the rear wave of the dipole speakers is lightly shaded to denote its negative polarity.

The two most important things to note about the polar response of monopole box speakers (shown on the top row) are the frequency-dependent increase in directivity that changes the forward dispersion from about 250Hz through the midrange and treble, and the spherical radiation pattern in the bass. A common misconception exists that rear-wave radiation is unique to dipole speakers. Fig.1 clearly illustrates that monopoles also have a very prominent rear-wave spanning the lower frequencies, but with the same polarity as the front. It's this change in directivity from omnidirectional in the bass to unidirectional in the higher bands that causes alteration in the overall sonic illumination of the listening room, emphasizing the bass region.

The second row of images shows the response of a bipolar speaker. As you can see, it has the same spherical bass-radiation pattern and low-frequency standing-wave problems as the monopole box designs, but the midrange and treble dispersions are fairly uniform, illuminating the room more evenly over a wide range of frequencies. However, the bipolar pattern still produces high levels of in-room reverberant energy.

The third group of images highlights planar dipole speakers. With a given on-axis spl, note the significantly lower levels of sound radiated to other parts of the room from all three frequency bands when compared to the other designs. Plus, the rear wave has reversed polarity, resulting in less low-frequency reinforcement. On the down side, the dispersion pattern becomes increasingly ragged at higher frequencies. These off-axis irregularities can generate overt colorations, degrade image quality, and increase speaker placement sensitivity. Fortunately, no radiation occurs 90° off-axis, reducing side-wall and ceiling reflections and lowering the strength of the overall reverberant field.

The final row illustrates how the Audio Artistry dipoles maintain a consistent dispersion pattern through the bass and midrange frequencies. This uniform dipolar directional radiation, particularly unusual in a subwoofer, minimizes bass reinforcement and standing-wave excitation. In addition, the increasingly directional radiation in the midrange and treble reduces overall reverberant energy.

Perhaps it will help to understand how dipole radiation reduces low-frequency room interactions if we imagine a pair of hypothetical Dvorak main panels in proximity to the nearest rear wall, side wall, and corner on one side of a room (fig.2). (For clarity, I'll ignore the effects of the other boundaries.)

The left and right dipole panels are labeled "L" and "R," and the listener position is at point A. If we substitute the rear and side walls with an imaginary mirror, a number of "phantom" image soundsources are created. Note that the dipole's positive- and negative-polarity lobes switch positions in the phantom sources. The negative lobes from the phantom sources representing the rear wall and corner tend to combine with the forward lobe of the actual speaker, resulting in cancellation or attenuation. Reflections from the side wall behave more like a traditional monopole except that the off-axis reflections are significantly reduced in amplitude, even nulling at 90°. Notice, too, that the nulled axis of the side-wall phantom speaker is aimed toward the listener. It's this mechanism that helps reduce the reverberant field level and the amount of side-wall (and ceiling)

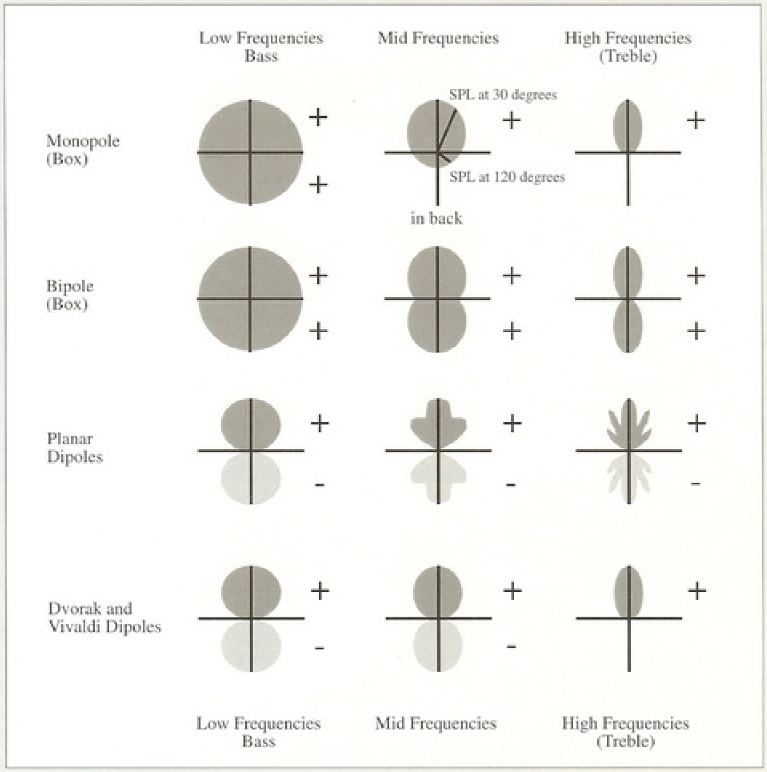


Fig.1 Typical loudspeaker polar patterns.

ably low levels of higher-frequency colorations. In addition, many such designs present a tough, reactive load to the amplifier. Large panels can also visually dominate a listening room, making their acceptance more challenging for the less audio-inclined members of a household.

Unfortunately, since almost all dipole speakers are also planars, the inherent positive attributes of dipole radiation are often mistakenly credited as being unique to panel speakers, while the negative characteristics that stem from the use of a physically large transducer are often falsely blamed on the dipole concept itself! If I had a buck for every time I've heard that "dipoles" are a bear to set up properly, I could easily buy a second pair of Dvoraks (which, by the way, are relatively easy to set up).

The best panel speakers share a number of positive qualities to greater or lesser degree: a crystal-clear and timbrally pure midrange, a freedom from boxy colorations, excellent transient response, and the ability to convey realistic image size. Dipoles also differ markedly from monopoles in a moderately directional radiation pattern over their effective bandwidth, and a reduction of low-frequency in-room reinforcement.

However, linear excursion capability is not among the intrinsic attributes of panel drive-units. Dipole panels that attempt to reproduce genuine bass require a very large transducer in order to move sufficient air; even then, the results aren't always satisfactory.

This need to move large amounts of air to reproduce the low-frequency foundation of music with convincing volume, dynamics, and definition is the fundamental problem with panel speakers. An unequalized dipole's response rolls off with decreasing frequency as the two opposite-polarity soundwaves increasingly cancel each other at frequencies where their separating path length is short compared to the radiated wavelengths. Progressively larger excursion is required, therefore, to maintain a constant sound-pressure level as the frequency drops. For example, for a dipole driver of any given size to generate a 50Hz tone at the same volume at the listening position as a 500Hz tone requires 1000 times the excursion.

Compare this to the performance of a conventional monopole woofer, which needs only 100 times the excursion to maintain the same volume at 50Hz as at 500Hz, and it's easy to see why dipoles put such serious demands on driver quality at their operating extremes. This limitation has given rise to many hybrid designs using conventional woofers to reproduce bass — with varying degrees of success. Yet taking this course means sacrificing the genuine dipole advantage of low-frequency directionality.

On the opposite end of the spectrum, panel transducers large enough to provide reasonable bass extension typically create a serious compromise in upper-midrange and treble reproduction. All speakers become more directional at

reflections focused toward him or her.

Referring back to fig.1, if you replace the dipole with a monopole, it should be easy to visualize how the omnidirectional, common-polarity reflections from all surfaces combine with the direct sound, reinforcing the bass.

Fig.3 illustrates the effect of reducing the overall reverberant energy relative to the direct on-axis sound-pressure level experienced by the listener with either a dipole or monopole. In the case of a dipole, you could sit 73% farther away from the speaker than with a typical monopole or bipolar speaker before the direct and reverberant fields blended! Consequently, while some acoustic treatment is still a good idea, you'll generally need less with dipoles than with box speakers.

—Shannon Dickson

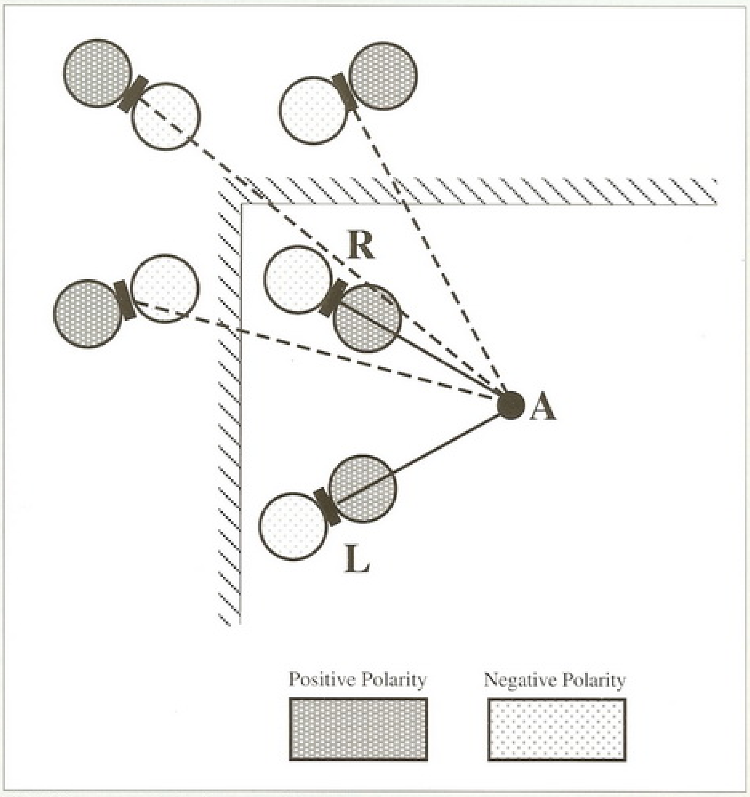


Fig.2 Mirror-imaged dipole room reflections.

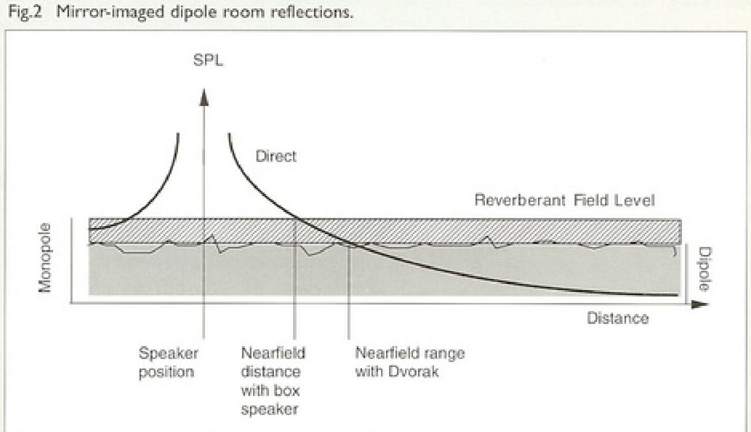


Fig.3 Comparison of direct sound *vs* reverberant energy for a full-range dipole speaker and a typical monopole box speaker.

higher frequencies, because the radiating area (including the driver and baffle dimensions) becomes equal to, then progressively larger than, the wavelengths of the sound. When the ratio of driver size to wavelength increases beyond a certain point, multiple lobes form in the radiation pattern, producing an uneven off-axis response (see polar-pattern diagrams in sidebar). Reflections from these irregular soundwaves can blend with the speaker's direct sound, adding coloration. The beamy treble also forces the listener to sit with his or her head in a virtual vise to get decent imaging.

These problems are potentially significant, considering the large radiating areas featured by many traditional dipole designs. Manufacturers like Martin-Logan, Sound-Lab, Magnepan, and Quad have developed clever partitioning schemes to minimize treble beaming and off-axis colorations in their designs. Nevertheless, these dispersion problems still exist to some degree with panel speakers.

Despite these very real limitations, the communicative powers of the best panel dipoles can be so stunning that their flaws are accepted, or at least overlooked. Once you've heard how such designs can portray music's scale and clarity, it's hard to do without it.

Moving-Coil Basics: Moving-coil box speakers dominate both high-end and consumer audio sales; the best examples are viewed as real-world benchmarks of excellence. However, no matter how well full-range moving-coils perform in an anechoic chamber — or a huge ballroom at a trade show — they're usually bought for use in moderate- to normal-sized living rooms.

When these speakers reproduce music with wavelengths significantly larger than their radiating surface and baffle dimensions, they radiate equally in all directions and excite many of the room's resonances. Since a wavelength of 50Hz extends approximately 22.5' and 20Hz is equivalent to 56.5', the drivers and cabinets of even the largest box speakers are tiny by comparison. As a result, all music below roughly 250Hz radiates from monopole woofers as omnidirectional, spherical soundwaves which then reflect off all adjacent boundaries in the listening room. As the wavelengths of these spherical soundwaves are generally larger than the distance from the speaker to the nearest boundaries, the reflections combine in-phase with the direct sound, resulting in a broad band of low-frequency reinforcement, the so-called "room gain."

In the worse case, up to 9dB can be added to the natural volume level in the bass, with peak-to-peak differences of around 20dB between the nodes and anti-nodes of specific room resonances! In practice, the actual frequencies, and the degree to which reinforcement or standing-wave excitation takes place, will depend on the speaker's low-frequency response, the specific absorption characteristics and dimensions of the room (including acoustic treatments), and where the speaker is placed with respect to the nearest corner and room boundaries.

The sonic result is often an unnatural fullness or, in severe cases, a turgid or bloated "one-note" character to low-frequency reproduction that obscures low-level detail and adds coloration clear up into the midrange. When this excess bass energy is combined with high levels of broad-band reverberant energy, often generated by cumulative reflections from monopole midrange drivers, overall resolution and dynamic contrast suffers. Omnidirectional bass can also skew the overall tonal balance of a recording because it conflicts with the increasingly directional dispersion of a speaker's midrange and treble.

Such problems may be lessened somewhat by careful low-frequency design and intelligent application of acoustic treatments. Use of room-placement computer programs — such as *Visual Ears*, described by JA in this issue's "Industry Update" — or a lot of educated trial and error, can also be of real help in reducing the amplitude of resonances excited by a conventional speaker. Those audiophiles who have taken the time to optimize speaker placement for a given room have no doubts about the importance this offers for increased sonic realism. However, even under near-ideal circumstances, genuine bass extension from a monopole speaker in an average-sized room can pose a significant barrier, curtailing the resolution of natural timbres and low-level decay that are captured on good recordings.

The Dvorak

The fundamental limitations faced by both dynamic box speakers and large planar designs underscore why Siegfried Linkwitz has spent the past nine years designing a speaker that attempts to minimize the weaknesses of both while building on the strengths of each.

Main Panels: During its years of development, the Dvorak underwent several iterations before maturing to its current five-piece configuration of two main panels, two subwoofers, and an active crossover/equalizer. While the overall panel is 56" high, the speaker proper consists of a folded MDF baffle measuring 26.5" H by 12" W by 4" D. This is supported about 20" above the base by a structural spine and two side legs running from top to bottom. These legs serve double duty as the folded portion of the baffle. On the rear of the main panel, three narrow horizontal boards complete the folded portion of the baffle. The rear spine, folded sides, and horizontal structural supports are all made from rigid Baltic Birch. A full-length black Crimpoline cloth covers the entire structure, and an Avonite top-piece puts the finishing touch on an appearance that's reminiscent either of a small Acoustat or a large RoomTune.

The folded sides of the baffle serve to minimize the frontal width of the panel, reducing edge diffraction while achieving the desired separation between the positive-polarity, forward-firing wave and the negative-polarity, rear-firing wave. Both Marshall and Siegfried emphasized that the precise shape and construction of the Dvorak's baffle were critical for maintaining uniform directivity and consistent dispersion characteristics across the frequency region where the transition between dipole and quasi-dipole/monopole radiation occurs.

Though free from typical cabinet resonances, the baffle does vibrate a little. However, as for the most part these vibrations also radiate in dipole fashion, Siegfried feels that effective cancellation of the front and back waves minimizes any audibility. Though the main panel is fairly rigid, I did notice a very subtle increase in focus by loading the rear of each base with the optional, $150/pair "Top Hat" — a black rectangular box filled with a 25-lb bag of lead shot.

Two Vifa 8" cone drivers are clamped in place on the baffle by the rear spine (no screws are used). These form a symmetrical array by being mounted above and below a 1" aluminum-dome tweeter, also from Vifa, which stands 41.5" from the floor. This tweeter is found in a number of well-respected high-end speakers, and was chosen for the Dvorak because of its excellent room response and power-handling capability around the rather low crossover point of 2kHz. Use of a dome tweeter means that the moderately directional radiation pattern of the dipole midrange-units and woofers is maintained through the treble — *sans* the rear wave — but with a uniform forward dispersion characteristic in lieu of the narrow, "hot-spot" beaming common with traditional dipoles. As a result, the speaker should avoid the placement sensitivity and off-axis colorations typical of some panel designs.

The dipole drivers cover a range of 40Hz to 2kHz when operated full-range without the optional subwoofers.

With the subwoofers, the active crossover rolls-off the 8" drivers at a more comfortable 100Hz. In addition, a soft absorbent material placed in the shallow cavity between these drivers and the grillecloth covering the back of the speaker partially absorbs and attenuates the upper-midrange portion of the rear wave. This helps achieve a smooth transition to the tweeter's forward-firing radiation pattern. (Linkwitz found dipole operation in the treble not beneficial, as it degrades overall tonal quality, produces high-frequency "splatter," and makes precise speaker placement more tricky.)

Driver selection is a critical feature in any speaker's performance, but particularly so for a dynamic dipole design. Without the assistance of an enclosure's loading properties, dipole radiation demands large excursion capabilities from a driver over portions of its frequency band as well as good power handling, inherently low distortion, and smooth cone break-up characteristics near the edge of its performance envelope. Excessive excursion can produce higher levels of inductive modulation distortion, degrading sonic clarity. By employing two 8" dipole drivers per channel instead of just one, more-than-adequate volume levels are possible in a normal-sized room without inordinate excursion. (Interestingly, even though they excelled in some performance areas, some of the more exotic high-ticket drivers evaluated by Audio Artistry didn't pass the grade for dipole use due to unacceptably high distortion when pressed near their operating limits.)

At the bottom of each main panel an 11" by 6" MDF board is braced vertically between the outside legs, further stiffening the structure against torsional vibration. This board also carries the passive crossover components, including Solen polypropylene capacitors, air-core inductors, and Caddock power resistors. The crossover is a 4th-order Linkwitz-Riley, with electrical compensation provided to minimize the time delay resulting from the physical offset of the drivers. Although the crossover point is a low 2kHz, the steep 24dB/octave filter slope makes life much easier on the tweeter. It also minimizes the consequences of any cone break-up artifacts produced when the 8" drivers are driven hard. A rigid aluminum plate on the lower portion of the center spine directly adjacent to the crossover contains two sets of gold-plated, multi-way binding posts to facilitate bi-wiring.

Dipole Subwoofers: The Dvorak's main panels sound fine when used alone in a small room, offering good dynamics and the essential open perspective of the full system. However, I strongly encourage adding the stereo subwoofers even if your room is not large.

A black grillecloth encircles the subwoofer enclosure, but the side-panels are made of solid MDF. The cabinet is coupled to the floor with adjustable spikes threaded into a 2"-thick plinth. A black Avonite top matches that of the the main panel. Each subwoofer contains two long-throw 12" woofers, each with a free-air resonance of 18Hz. By using equalization to flatten the natural dipole rolloff, you can get LF extension with adequate power through the full audible range.

The top woofer faces forward, the bottom woofer rearward. Both radiate sound in a dipole pattern. In order to reduce the overall width of the subwoofer cabinet, the drivers are mounted on angled baffles, one atop the other and criss-crossing at 45°, and separated by a horizontal board that runs through the center of the enclosure. The dimensions of the enclosure, as well as of the space between the driver cones and the cabinet openings, are calculated to enhance the drive-unit sensitivity. The potential cavity resonance formed by the internal space is much higher than the 100Hz maximum frequency the subwoofer is intended to reproduce, so it should not create a problem.

A pair of gold-plated binding posts attached to the lower rear of the cabinet connects the amplifier to the big woofers. This doesn't waste amplifier power in a passive crossover. In addition, the amplifier directly damps the woofers, providing superb control. A 50Wpc amplifier is more than sufficient to use with the Dvorak's subwoofers, as it only takes about 25W to drive the big woofers to their peak excursion.

Active Crossover: Even if you don't purchase the Dvorak's optional dipole subwoofers, the external active crossover/equalizer is employed to extend the low-frequency rolloff of the main panels to 40Hz. With the subwoofers in the system, however, the crossover between the main panels and the big woofers is set to 100Hz. This significantly reduces the excursion demands on the 8" drivers, allowing them to operate in a more linear fashion. Specific equalization is also applied to both the feed to the main-panel amplifier and that for the subwoofer. This equalization compensates for the natural dipole 6dB/octave rolloff below 100Hz for the subwoofer's 12" drivers, and above 100Hz for the rolloff of the 8" drivers. (This rolloff occurs for all frequencies where a half-wavelength is greater than the separation between the center of the front and rear radiating elements.)

The active crossover is contained in a black sheetmetal enclosure. A blue LED shines from a ½"-thick, black-anodized aluminum faceplate. One input and two output RCA jacks per channel are mounted on its rear panel. Push-button switches allow each main panel and subwoofer to be turned on or off independently. If you don't buy the subwoofers, you simply turn the crossover's woofer circuit off and connect the pair of main outputs to a single stereo amplifier. Otherwise, two stereo amplifiers are used.

There is also a separate woofer-level adjustment control knob for each channel. These potentiometers provide a 12dB range to precisely match the system's bass response to a given room and/or balance any inherent level differences between the two amplifiers — should there be a difference. A front-mounted "video" button prevents the subwoofers from bottoming-out when playing the occasional excessive soundtrack explosion by adding a gentle, 6dB/octave high-pass attenuation from around 50Hz downward.

Inside the active crossover a single circuit board occupies the chassis' full width and more than half its depth. This board contains an additional ±12dB bass-level switch that, along with the external woofer-attenuation knob, provides as much flexibility as one could ask for in optimizing low-frequency performance. While the design of the electronic crossover is clean and straightforward, the Dvorak is built to a relatively affordable price point, so you won't find a "Who's Who" list of audiophile-approved passive parts. On the other hand, circuit design—by far the most important consideration in determining overall performance of electronic gear—is well executed, and the parts used in the Dvorak crossover are of good quality.

Inputs and outputs are AC-coupled using polypropylene capacitors. Along with 1% metal-film resistors, the same type of caps are used to implement the filter and EQ circuits. Four op-amps are used for each channel, these the excellent Burr-Brown OPA-2604. Burr-Brown dedicates a whole page of their data sheet to a description of the special sonic qualities of this modern FET-input op-amp — an unusual step for a mainstream semiconductor company. The OPA-2604 is used with carefully optimized support circuitry in some of the best-sounding high-end gear in today's market, such as the Mark Levinson No.38S preamp.

Audio Artistry has developed an ultra-quiet balanced crossover to accommodate their new flagship Beethoven speaker. A revised version of the crossover for the Dvorak incorporating some of the same refinements will likely follow in the near future. One of the nice things about an external active crossover is how much easier it is to upgrade: you don't have to send the entire speaker back to the factory to take advantage of inevitable progress.

The 180mA, ±12V "wall-wart" power supply shipped with the Dvorak's crossover was selected to keep the speaker affordable and to avoid hassles when exporting the system to foreign markets, not to provide ultimate performance. Alternative power supplies are available that may provide some subtle refinements, yet the stock supply is certainly adequate. Also, keep in mind that the sonic impressions rendered in this review were based largely on the stock supply.

With a retail price of $5990 for the entire five-piece system, the Dvorak is reasonably priced. (The Dvorak subwoofers are also available with an unequalized version of the crossover for use with other speakers that could use the benefits of dipole bass. Price: $3590.)

Setup

The Dvoraks may be less sensitive to room placement than panel dipoles and most box speakers, but some care and experimentation will certainly pay off. I recommend placing the speakers at least 3' to 4' from the back wall, and a minimum of 2' from the side walls. Toe-in the main panels so that the tweeters are nearly on-axis with your ears, then place the two subwoofer cabinets slightly in front and on the outer side of each main panel so that the center point of each subwoofer cabinet is the same distance as the nearest tweeter from your respective ear.

This time-aligns the woofer voice-coils with those in the main panels with respect to your ears. You can experiment by sliding the woofers along an imaginary arc that connects the center of the two cabinets and intersects the tweeters in both panels. The ideal spot for each will vary with your room's dimensions and your sonic preferences. If you have the space, try a tweeter-to-tweeter spread of 8' or more. And be sure to carefully level the panels from left to right and from front to back.

Those who use speaker-placement software or formulas for smoothing the bass response in a given room should keep in mind that, where an omnidirectional woofer will give its smoothest response when located at the nodes of a room's modal profile, a dipole is just the opposite with respect to the front-to-back axial modes. The relationship of side-wall and ceiling axial modes is similar for the two types, but because dipoles have a null at 90°, less off-axis interaction will occur in these two planes, particularly if the speakers are at least a couple of feet from the walls. Therefore, if you wish to further smooth the Dvorak's bass response, try placing the panels and subs in the anti-node nearest the node you would normally choose to place a box speaker.

Whether the final woofer placement is nearer the panels or the side walls, I do suggest you toe them in a bit as well—though not necessarily on-axis like the main panels. Some degree of toe-in, however, will give you increased focus from the directional bass. It's also possible to move your listening position much farther back than you would with most box speakers and still get good imaging. This is due to the reduced amplitude of the overall reverberant field and the weak side-wall/ceiling reflections which, in turn, expand the "near-field" response window compared with a monopole.

Watch the woofer level: Though the marriage between the dipole woofers and main panels is seamless, as the balance is partially in the hands of the user, the flexibility afforded by the wide range of bass-level adjustments makes defining a specific sonic character for the Dvorak's bass and midrange response difficult. The most common mistake people make with the speaker is to turn the woofer levels up too high—they misinterpret the resultant thumping quality as "good" bass. Don't do it! The owner's manual gives a thorough guideline for dialing-in the woofers for your particular room and associated amplifiers, so I'll just highlight a few tips. It's not difficult to do, but it does take a little time and is best accomplished with the help of a friend.

When following the woofer-adjustment instructions, pay attention to the timbre or tonal character. If the balance gets leaner with the woofers in the circuit, turn the level up on the crossover. Be careful, because adjusting the level knob by one "hour" changes the bass level by 1.2dB; even a small step in low-frequency output can produce a surprisingly big sonic change.

If the sound becomes thicker, more "tubby," with the woofers on, turn it down. With the bass levels set too high, dynamics are curtailed, midrange focus and definition degraded, kick drums will sound as if the drumskins are damped with pillows, and bass guitars can sound like someone's playing with leather gloves on.

When you've got it right, the system's overall tonal balance should be the same with or without the woofers engaged. However, with the subwoofers, the midrange has greater presence, the focus improves, and the soundstage is better defined. The effect can be awe-inspiring: bass is conveyed with a correct mix of transient snap and weight while improving resolution of midrange harmonics. In addition, you simply cannot hear the woofers as a discrete sound source. I've found that the final woofer-level adjustment will range between 11:00am and 1:30pm for most average-sized rooms when both amplifiers have the same amount of gain. The best setting may differ in larger or smaller rooms or when dissimilar amps are used.

The big picture

The Dvorak is the first truly full-range speaker I know of that appears to maintain a moderately directional radiation pattern throughout the entire audible bandwidth, yet whose radiating surfaces are relatively small compared to the wavelengths they produce from 20Hz all the way up through the treble. As a result, the dominant signature of the listening room is reduced by an unprecedented amount in a dynamic speaker, allowing the listener to experience a literal wealth of low-level detail, textural purity, and natural timbres. One of the principal reasons for this outstanding expressiveness is that the Dvoraks fill a listening room with two-thirds less reverberant energy for a given on-axis volume level compared to a pair of conventional box speakers! In other words, substantially more sound is focused toward the listener and less is radiated to other areas of the room. By the way, the Gradient Revolution shares some of these characteristics, as does the Quad ESL-63 when partnered with the Gradient SW-63—another dynamic dipole subwoofer—but neither are true full-range designs.[3]

My usual listening experience was transformed to a perspective I found more natural and engaging than that produced by any other speaker I've heard at length—to date, that is. I'm not claiming that the Dvorak is superior to all other speakers—it's not. But it demonstrates that by applying known principles in a novel manner, several serious limitations that are consciously (or perhaps subconsciously) taken for granted can be dramatically reduced. I also feel that the Dvorak's fundamentally more natural presentation is more important than many of the desirable sonic attributes on the typical audiophile checklist, even though the speak-

3 A few other companies offer dynamic dipole speakers. Genesis has the models V and VI, Legacy has their Whisper, and the granddaddies of them all are the Celestion System 6000 and the Enigma panel subwoofer. —JA

er does perform well when assessed by these more traditional standards.

The Dvorak also illustrates an interesting paradox I've noticed with other outstanding components. A product that combines a special blend of attributes that improve overall fidelity, rather than just providing specific sonic changes or enhancements, seems to get out of the way of the performance. Such components not only are more resolving but allow the listener to more easily separate uncorrelated distortions and anomalies from the music, making them far less distracting. With other gear that lacks these special qualities, but is still considered revealing, irregularities either embedded in the recording or generated elsewhere along the reproduction chain normally bother the listener more because they are intertwined with the music.

I've also seen an interesting array of reactions from various people after their first exposure to the Dvoraks. While nearly everyone noticed their captivating viewpoint from the outset, a couple of audiophile friends felt a little puzzled after the first session with the speakers —almost as if something were missing. They were missing the "room." The contribution of a room's reverberant field and standing waves is so ingrained in our experience of reproduced music that many of us take it for granted, however much we may acknowledge its impact intellectually.

This is an important point, and one common to many areas of subjective experience. A loose analogy can be drawn comparing the various distortions heard with LP playback: in addition to its numerous inherent positive attributes, many of us have learned to prefer the LP through years of acclimation. Accordingly, constant listening to speakers with omnidirectional bass can lead to a conditioned preference for overpowering but dynamically constricted bass impact, or excessive low-frequency "slam" that is neither on the recording nor representative of the real thing. Once the Dvoraks were placed in my system, I went through a few days of adjustment as the lack of low-frequency "room compression" I'd become accustomed to all these years was replaced with an open, effortless, and full-bodied bass reproduction.

The Dvorak Difference

Right from the start, the Dvorak's startling dynamics of voices and instruments was apparent, as was an alluring resolution of low-frequency ambient information and subtle textural detail which (I surmise) is normally masked in my room. Surprisingly, the most pronounced effect of the clean bass performance was greater clarity and presence throughout the midrange. With the bass as extended, tactile, and well-integrated with the rest of the spectrum as it was with these speakers, even ordinary recordings became more interesting, more involving.

With the Dvorak's better perceived dynamic contrast, correct playback volumes of my favorite recordings varied over a wider range. Fine gradations in volume setting were more easily appreciated, making me value all the more the Rowland Coherence preamp's remote control and awesome transparency. Though the midrange perspective reminded me of the Quad '63's, with the Dvorak this desirable quality extended all the way down through the subterranean foundation of music. It's difficult to overestimate how important open, articulate deep-bass reproduction is in allowing your mind to suspend its disbelief and become engrossed in the music when devoid of the typical veiling colorations and overhang from resonant artifacts.

In particular, I noticed two important qualities that define this characteristic. First, the room doesn't take off on its own when its resonances are excited. While this room "compression" can make the impact of large crescendos or high-intensity transients seem visceral, certainly increasing the "Wow!" factor, much of its impact comes from reverberation and resonant feedback rather than from the direct natural energy of the music. Dynamic contrast becomes restricted, analogous to the way in which electronically compressed music allows a loud but shallow range of expression.

Subjectively, this room distortion feels as if the sound is pressing in on you from all sides during a high-level transient. By contrast, the Dvoraks allow you to hear the swing from quiet to very loud and back with remarkable fidelity. Listening to well-made LPs like *The Fantasy Film World of Bernard Herrmann* (Mobile Fidelity MFSL 1-240), I had the sense of sharing the recording's original acoustic environment. A musical swell or crescendo would extend from the soundstage and envelop me without the artificial pressure from excessive room-mode excitation. Be forewarned: Unless you have a very large room, it's difficult to go back to traditional speakers, even those of outstanding design, after adapting to the Dvorak's realistic and appealing presentation.

Second, because of the reduced amount of resonant overhang and lower overall reverberant energy in the room, the decay of musical passages and notes is simply more lifelike. Fine nuances and ambient cues normally buried in the room's noise floor are revealed. The ability of these speakers to clearly dis-

Reference System

I used a wide variety of gear and recordings while evaluating the Dvorak:

Front end: Immedia RPM-2 turntable and arm with Lyra Da Capo cartridge; Theta Data II transport with Muse Model Two, Theta DS Pro Generation V-A, and Sonic Frontiers SFD-2 Mk.II processors; Rowland's Consummate phonostage and Sonic Frontier's SFP-1 phono-amp.

Amplification: the outstanding Rowland Coherence preamp and two pairs of Model 6 monoblock amplifiers, all battery-powered/transformer-coupled, were my primary references; Sonic Frontier's SFL-2 and the exciting new Audio Research LS22 provided alternate preamp duties; Ayre V-3 and BEL 1001 Mk.II power amps also proved valuable references.

Comparison Speakers: Avalon Radians.

Cables: digital cables were Cardas AES/EBU, Marigo Apparition Reference, Aural Symphonic Statement, and AudioQuest's Opti-link 2-ST; Cardas Golden Cross and Discovery balanced interconnects were used, as were Cardas Golden 5-C, MIT CVT 770 and 850 Terminator, and TARA Labs RSC Decade speaker cables.

Accessories: a Vibraplane or a pair of custom Newport pneumatic platforms isolated my transport and turntable; Townshend Seismic Sinks supported all other gear with D'Feet SH-22 pucks in between; Mike Fredericksen equipment stands were used, as were Audio Power Industries' reliable 116, 112, and 110 Power Wedges and Power Enhancer. API and Cardas power cords delivered AC to all gear requiring it; ASC Tube Traps helped acoustically; a VPI record cleaner, several Bright Star Little Rocks, and some Shakti "Stones" rounded out the field. —Shannon Dickson tinguish various recording styles is remarkable. Microphone types and arrangement, as well as the quality of multitrack mixing and relative distances between elements within a naturally recorded soundfield, are more easily delineated with the Dvorak than when it is overshadowed by room compression. Even with well-known recordings, I'm still surprised by the wide-band low-level resolution that was swamped by acoustic compression with other excellent speakers in my 17' by 23' by 8' room.

The combination of these two major attributes produces the Dvorak's open-sounding perspective. With the right material, boundaries really did seem to disappear, allowing the acoustic soundscape to permeate the room and sonic images to pass right through the speakers rather than bend around them. The Dvoraks convey the scale, intensity, and presence of a real piano, tenor sax, or lead vocalist better than any speaker I've heard. One listen to Johnny Hartman's elixir-like voice on *John Coltrane and Johnny Hartman* (Impulse! AS-40, reissue GR-157), and I suspect it will be "case closed" — these speakers portray voices so beautifully!

I've heard outstanding moving-coil speakers that can focus images like nobody's business, defining pinpoint, almost holographically defined instruments in width and depth. However, before I auditioned the Dvoraks, I'd yet to hear one that could also convey the height of instruments, musicians, or the recorded ambient environment in proper proportion to the other two planes. By "proper proportion" I'm referring not only to the dimensional aspects of an acoustic image, but also to the intensity or "body" with which it is projected. With the Dvoraks, the listener seems to breathe the same air as the performers: It's as if the space between your chair, the side walls, and the rear boundary of the recording venue were illuminated with a vibrant, real-time presence. Listening just now to Classic Records' awesome new 45rpm re-reissue of Mussorgsky's *Pictures at an Exhibition* (RCA LSC-2201), I felt as if I'd been shot with a stun gun on my way through the Great Gate of Kiev.

In light of the total experience conveyed by the Dvoraks, the slightly skewed perspective of less-open-sounding but otherwise excellent moving-coil speakers always reminded me that I was listening to a loudspeaker — no matter how precise, beautiful, and engaging their timbres and tonal balance, or how sharply defined their imaging. Without the usual, more obvious box-speaker distortions induced by the cabinet and air-cavity resonances radiating back through the drivers, the "character" of the Dvorak's sound is dominated by a seamless soundscape populated with full-bodied imaging. The presentation of the Dvoraks is closer to the live experience than that of speakers that can carve perfect sonic outlines within a multilayered soundstage but lack the Audio Artistry's sense of scale and presence. Several electrostatic and magnetic planar models can get image scale largely right, and do so with excellent transparency and clarity. However, I've yet to hear one of these models that also possesses the dynamic contrast, natural timbre, and low-frequency authority of which the Dvoraks are capable.

Another major plus that stems from the Dvorak's diminished room excitation is how "apartment-friendly" this full-range speaker is. You can rock out with +90dB levels at your listening chair, yet the walls vibrate far less than with a full-range box speaker. While standing in the hallway outside the closed door of my apartment, I could just discern that music was playing with Dean Peer's bass-guitar tone poem *Ucross* (Redstone RR-91012) blasting at high volume!

Characteristics & caveats

Now that I've shared my impressions of the Dvorak's most prominent qualities, it's worth looking at several other distinguishing characteristics and a few areas for possible improvement.

The Dvorak's upper-midrange and treble performance, though slightly shy of the state of the art, is competitive with many of the most respected high-end designs. In the past, I've been particularly sensitive to the hardness typical of some metal-dome designs. Fortunately, the Dvoraks join a growing list of high-quality speakers that prove that modern aluminum and titanium tweeters, when mated to good crossovers and associated components, can fill the bill in fine fashion. I noticed no distracting high-frequency anomalies, and the tweeter sounded relatively smooth and extended — though, as I implied earlier, I've heard a few very expensive designs with a shade more treble transparency and fine inner focus.

The Dvorak's treble "flavor" leaned more toward "clear and detailed" than "warm and sweet." Treble artifacts arising elsewhere in the recording or playback chain may become apparent. If you have a moving-coil cartridge with a rising top end, for example, or an edgy-sounding CD player, the Dvorak will let you know it, but without the bite or glare heard from tweeters and crossovers of lower quality. With decent source material, I never found the Dvorak's treble fatiguing or harsh in my system. However, like most speakers, it needed a good break-in period; be patient for the first 75 hours or so.

The integration of the Dvorak's 8" drivers with the tweeter and the separate woofers was first-rate — the excellent overall tonal balance and resolution didn't leave me pining for more. But if you run the main panels without the subwoofers, you can hear increased distortion when playing certain high-intensity, low-frequency material. Hence, my recommendation to use the subs, even with smaller rooms.

With the Dvorak's subwoofers you get both quality and extension, but even the four 12" drivers had limits on how loud they could play the lowest frequencies. In most rooms you'll probably get more than adequate levels of the deepest bass found on most recordings. However, there are a small number of CDs, such as *Pomp and Pipes* (Reference Recordings RR-58CD) that have an ultra-loud deep-bass transient or two that can bottom-out these woofers. Engaging the gentle rolloff with the "Video" button will allow you to play nearly anything without hitting the woofer stops. If you have a particularly large room or a penchant for maximum-volume video soundtracks, you could spring for Audio Artistry's larger subwoofers, each containing four 12" drivers to give lower-distortion, higher-output bass reproduction.

These comments about the Dvorak's bass performance are made in comparison to other loudspeakers with which I'm familiar. I certainly don't mean to imply that it reproduces this most difficult region of the audible band with complete fidelity, even when optimized. No speaker I know of is free from noticeable low-frequency distortions, and the Dvorak is no exception. But it is capable of providing a more convincing and fulfilling replica of the qualities I hear from real bass instruments in a typical room than any other speaker I know of.

Learning curve

Any new enterprise will experience its share of start-up manufacturing glitches, regardless of how skilled the principals are at designing or business planning. Accordingly, I had my eye out for any rough edges in build quality or packaging of the Dvoraks. After a year of putting the speakers through some very tough paces, only three such problems cropped up, all of which have been solved for current production models.

First, the original shipping box left a little to be desired. It was reasonable for across-state ground freight, but by the time the speakers made it from North Carolina to Hawaii they had a few dings on the woofer base, and the outer box was a bit banged-up. Audio Artistry now ships all Dvoraks in a very-heavy-gauge, double-boxed crate with excellent inner shock-absorbing material.

The second problem occurred with the original binding-post plate. It was rather thin, and the insulating washer that prevents the binding post from electrically shorting with the plate was not well seated on my review samples. Consequently, the binding-post nut loosened on one speaker, shorting the amplifier. New, thicker aluminum support plates have been installed, the insulators are seated in a larger hole, and "lock-tight" prevents the nuts from coming loose.

Finally, after some use I began noticing a buzz emanating from one speaker whenever a signal with strong 80Hz content was played. I first thought the driver's voice-coil was rubbing, but when I sent the panel back to Audio Artistry, they discovered that the cloth around this panel was a little too tight around the main baffle: a heavy 80Hz note would excite a resonance in the grillecloth. The problem was fixed by slightly offsetting the cloth from the baffle.

All things considered, I was impressed not only by how few problems I encountered after a full year with a novel speaker from a new company, but also by the prompt, professional manner with which each problem was addressed. —**Shannon Dickson**

Measurements from JA

The Dvorak's calculated sensitivity was a little higher than the specification, at 90dB/W/m (B-weighted). The main panel's electrical impedance (fig.1) showed the specified minimum of 3.1 ohms at 2.1kHz; though the magnitude is mainly much higher than this, note the high phase angle in the low treble, which means that the Dvorak will be quite a hard load for an amplifier to drive. In addition, the large variation in magnitude means that the speaker's frequency response will vary significantly if an amplifier with a high output impedance is used. In particular, the entire treble region will be shelved down—a tube amplifier will definitely make the Dvorak sound sweeter, as long as it is not fazed by the awkward loading between 1kHz and 3kHz.

As there is no cabinet to speak of, the low-frequency peak in fig.1 is not due to the usual box resonance. Rather, it is due to the free-air resonance of the mid/woofers, which appears to lie at 31Hz. The little wrinkle in the traces at 26kHz is due to the metal-dome tweeter's "oil-can" resonance.

Fig.2 shows the dipole subwoofer's impedance magnitude and phase. The peak below 20Hz is due to the drive-units' free-air resonance; the rise in magnitude in the midrange is due to the drivers' voice-coil inductance. But note that there are two strong wrinkles in the traces, one at 1500Hz and one around 180Hz. These will be due to resonances of some kind. While the higher-frequency one is innocuous, the lower-frequency one is less than an octave above the crossover to the main panel—risky business, unless the low-pass crossover filter is very steep-sloped.

The shaped output responses of the Dvorak's electronic crossover are shown in fig.3. The input level was 100mV and the subwoofer level control was set to its maximum position. The subwoofer drive signals, shown to the left of the graph, reveal a combination of cut above 100Hz and boost below to compensate for the dipole rolloff. The subwoofer drive is down by almost 20dB at the resonance frequency, which should result in minimal excitation of the problem. The lower subwoofer curve shows the effect of the "Video" switch: it cut the boost with eventually a 6dB/octave rate to avoid overloading the drive-units with subsonic explosions.

The main-panel drive signals are shown to the right of fig.3. The signal is increasingly boosted below 300Hz, again to compensate for the speaker's dipole rolloff. With the high-pass crossover switched in (bottom curve), the boost is reduced below 100Hz.

Electrically, the crossover performed well. Its input impedance (at 1kHz) measured just under 20k ohms, while its output impedance from the main outputs was 235 ohms (panel outputs at 1kHz) and 225 ohms (subwoofer outputs at 100Hz). Its insertion loss of 0.4dB will not be significant, and its distortion and noise were both low. Fig.4 shows its overload points: The top trace shows that the panel output at 1kHz clips at an output of 6.2V, equivalent to an input voltage of just over 7V, which is well above the maximum output level of any combination of source components and preamplifier with which the Dvorak will be used. The bottom trace is the subwoofer performance, assessed at 100Hz with the level control full up: The lower distortion level is due to the output's low-pass function, any additional harmonics being rolled-off; the output clipping point (1% THD+N) is 4V, equivalent to an input of 11V, this again well in the safety region. Though the low-frequency boosts applied to both sets of output signals will reduce the clipping margin, this should not be a practical problem.

As Siegfried Linkwitz points out in the accompanying interview, assessing the low-frequency performance of dipoles is not easy, due to the usual nearfield techniques failing to allow for the dipole cancellation. For interest's sake, however, fig.5 shows the nearfield output of the Dvorak subwoofer, measured without the crossover/equalizer (top trace) and with the crossover set to normal operation (middle trace) and to "Video" (bottom trace). Without EQ,

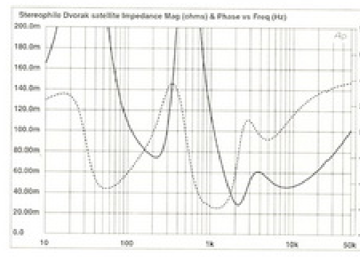


Fig.1 Audio Artistry Dvorak main panel, electrical impedance (solid) and phase (dashed) (2 ohms/vertical div.).

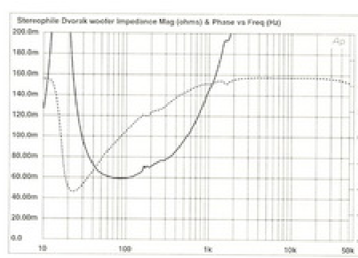


Fig.2 Audio Artistry Dvorak subwoofer, electrical impedance (solid) and phase (dashed) (2 ohms/vertical div.).

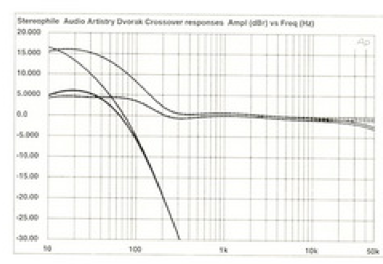


Fig.3 Audio Artistry Dvorak electronic crossover, responses of (from top to bottom at 70Hz): main panel outputs, subwoofer outputs switched off; main panel outputs, subwoofer active; subwoofer outputs (normal); subwoofer outputs ("Video") (5dB/vertical div., right channel dashed).

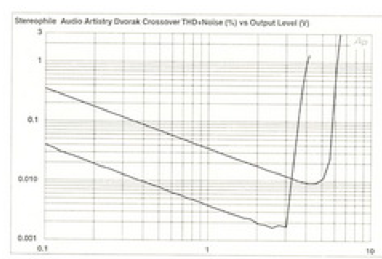


Fig.4 Audio Artistry Dvorak electronic crossover, THD+noise (%) *vs* output voltage into 100k ohms: main outputs at 1kHz (top); subwoofer outputs at 100Hz (bottom).

the subwoofer's natural output extends to above 1kHz. Despite the impedance wrinkle at 180Hz, the highest acoustic output is obtained at 270Hz. Note, however, that this peak is knocked down by 40dB once the EQ is switched in! It should not have any audible effect. Note also that the overall LF boost appears to be very mild, at 3dB or less (the difference between the levels below the 20Hz region with and without EQ). With the "Video" button pressed in, the entire 30–50Hz region is reduced in level by about 5dB, reaching –12dB at 20Hz, significantly reducing the excursion demands on the drive-units.

The individual responses of the main panel's drive-units are shown in fig.6. (Again, note that the woofer's nearfield traces are not representative in that they don't allow for the dipole cancellation.) The acoustic crossover is to spec at 2kHz, with steep, approximately 24dB/octave slopes. The passbands of the mid/woofer and tweeter look impressively flat. However, there does appear to be a little too much overlap between the drive-units to give a perfectly flat summed response through the crossover region. In my experience, the steeper the crossover slopes, the closer the tolerance necessary of the parts used if the actual crossover is not to depart from the target performance.

Note the deep, narrow notch at 24kHz. This is at its most extreme exactly on-axis and is due, I imagine, to the "phase plate" that covers the tweeter dome. The middle trace on the left of fig.6 shows the nearfield response of one of the mid/woofers, while the top trace is the same woofer's output modified by the electronic crossover. Remember, however, that these measurements do not show the rolloff due to the dipole cancellation.

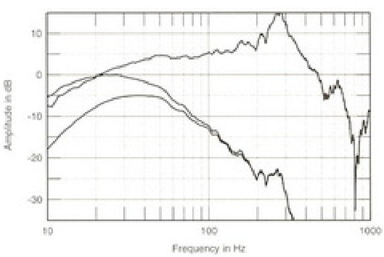


Fig.5 Audio Artistry Dvorak subwoofer, nearfield responses (from top to bottom): without EQ; with EQ/crossover (normal); with EQ ("Video").

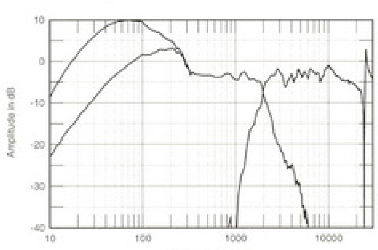


Fig.6 Audio Artistry Dvorak main panel, acoustic crossover on tweeter axis at 50", corrected for microphone response, with nearfield woofer response plotted with/without subwoofer crossover EQ below 350Hz. The actual cancellation due to the rear wave is not accounted for in the nearfield measurements below 350Hz.

The Dvorak's main-panel overall response, measured on its tweeter axis at a distance of 50" and averaged across a 30° horizontal window, is shown in fig.7. The top audio octave appears to be rolled-off in this graph, partly due to the tweeter being quite directional in this region. While the midrange and treble regions are otherwise impressively flat, the 1.5–3kHz crossover region is plateau'd up by 2dB. Whether or not this is due to the drive-unit overlap in this region, I could hear it in my own auditioning as a narrow band of brightness. Note also that Shannon did find the Dvorak to sound a little on the analytical side, which is not unexpected given this kind of on-axis balance (though the speaker's off-axis behavior is also relevant here).

On the left of fig.7 is the same equalized nearfield woofer response as shown in fig.6, plus the same measurement made with the high-pass crossover filter switched-in. To get an idea of the Dvorak panel's true bass extension, I used noise with a 1kHz bandwidth and measured the speaker's in-room response at the same 50" distance used to obtain the traces to the right of figs.6 and 7. Plotted on a half-octave-smoothed basis to eliminate some of the major room effects, this is shown as the bottom trace in the graph. The dipole cancellation does appear to cancel almost all the nearfield bass boost. However, there is still a slight energy excess apparent in the lower midrange. I assume that as SD didn't remark on any coloration in this region, its effect is benign, particularly as it falls in a spectral region where room effects dominate the perceived balance.

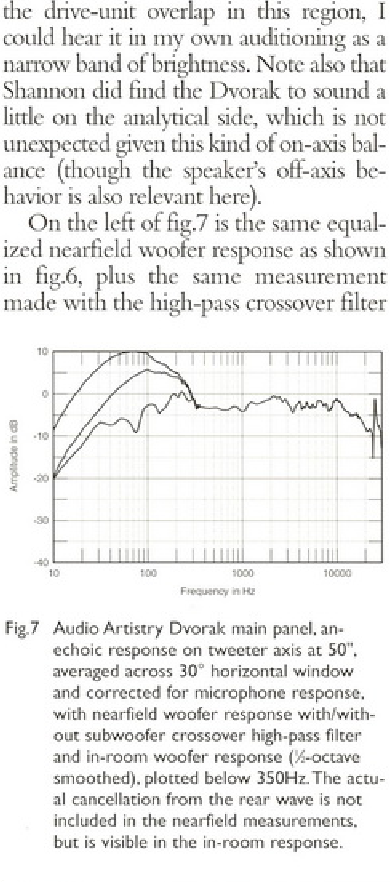


Fig.7 Audio Artistry Dvorak main panel, anechoic response on tweeter axis at 50", averaged across 30° horizontal window and corrected for microphone response, with nearfield woofer response with/without subwoofer crossover high-pass filter and in-room woofer response (½-octave smoothed), plotted below 350Hz. The actual cancellation from the rear wave is not included in the nearfield measurements, but is visible in the in-room response.

Vertically, while the tweeter's 41.5" distance from the floor is a little elevated for typical seated ear heights, the Dvorak's balance remained remarkably even across a wide range of listening heights. As long as the listener sits with his or her ears between the bottom of the bottom mid/woofer (35" from the floor) and the top of the top mid/woofer (51"), there should be no significant changes in balance. Only a standing listener close to the panel will hear a response with a lack of energy in the crossover region.

Horizontally (fig.8), the Dvorak's output falls relatively evenly with increasing off-axis angle, as expected. In the bottom octave of the tweeter's bandpass, where the dispersion is at its widest, the output at 90° has fallen significantly more than at 45° off-axis. However, the null at the speaker sides is less deep than I had anticipated, something that could be heard as well as measured. The front and rear waves do not see sufficiently the same acoustic environments to cancel completely at 90°. According to Siegfried Linkwitz, they cancel at around 110°, and the overall total power output follows the classic dipole characteristic.

In the time domain, the Dvorak panel's step response (fig.9) indicates that the tweeter and mid/woofers are connected with opposite acoustic polarity.

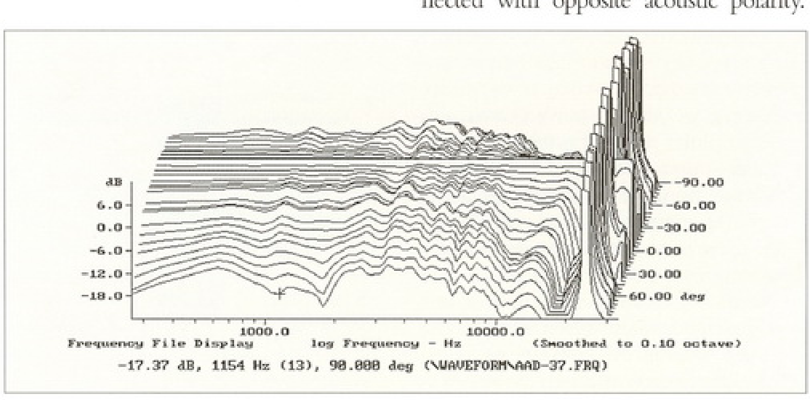


Fig.8 Audio Artistry Dvorak main panel, horizontal response family at 50", normalized to response on tweeter axis, from back to front: differences in response 90°–5° off-axis; reference response; differences in response 5°–90° off-axis. Note that the apparent peak above 20kHz is due to an on-axis notch "filling in" to the sides.

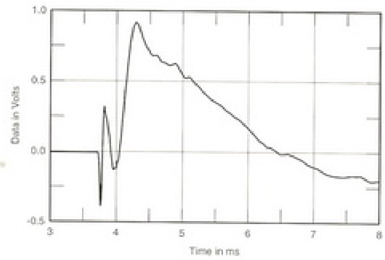


Fig.9 Audio Artistry Dvorak main panel, step response on tweeter axis at 50" (5ms time window, 30kHz bandwidth).

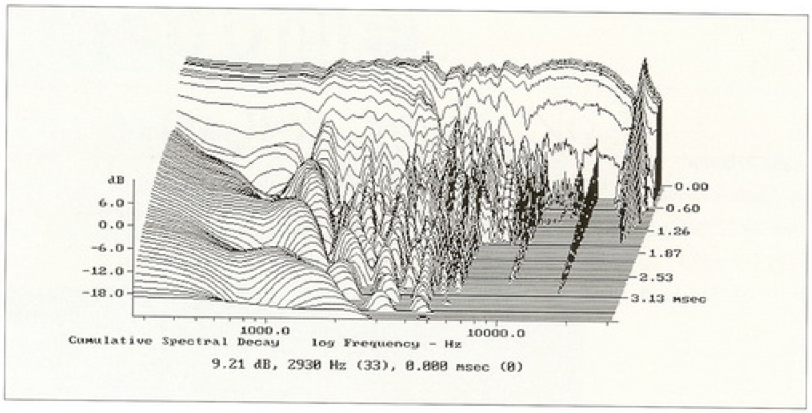


Fig.10 Audio Artistry Dvorak main panel, cumulative spectral-decay plot at 50" (0.15ms risetime).

(The tweeter's output is the small, initially negative-going spike of energy at 3.75ms; the woofers' output is the larger, positive-going triangle of energy immediately after the 4ms mark. As Linkwitz states in his interview, whether such a lack of time coherence is significant or not is not known. However, my experience is that if everything else in a speaker design is right, time coherence adds that last, essential element of image focus.)

Finally, the Dvorak's cumulative spectral-decay plot (fig.10) indicates an initially clean decay, but with some low-level hash present in the mid-treble. This may be resonant behavior; alternatively, and in my view more probably, this is due to low-level early reflections of the sound from the speaker's structure. Note that the ultrasonic tweeter resonance is down in level in fig.10, which was taken on-axis. The resonance is more noticeable—at least to a microphone—off-axis. —**John Atkinson**

Summary thoughts

After a year of careful listening, my take on the Audio Artistry Dvorak is that it successfully incorporates many of the positive qualities of large panel dipoles and moving-coil designs while minimizing most of their respective drawbacks. In contrast to so many speakers that are simply variations on common themes, the Audio Artistry Dvorak offers a refreshing perspective. With its marked lack of room, cabinet, and air-cavity colorations, its smooth and consistent power response, its natural top-to-bottom tonal balance, and its unforced yet dynamic expressiveness, I felt that I was hearing all that this combination of components could offer. Its wonderful balance of attributes answers the question posed at the beginning of this review—"Does this speaker really possess a unique and compelling communicative skill?"—with a resounding "Yes!" The Dvorak is a genuine music-lover's speaker of the first order.

While the Dvorak scores highly on the typical audiophile sonic checklist, its ability to tie the whole experience together and directly convey the emotional undercurrent of good music made conducting such an analytical assessment seem almost frivolous. Yes, a very few of the many outstanding speakers I have heard at length have a slight performance edge in a few specific areas, but I wouldn't trade the Dvoraks for any of them for sheer musical enjoyment or long-term reference in a normal-sized room. This includes a number of elite models costing nearly three times the Dvorak's $5995 price! Given unlimited resources and a cavernous room—well, since I live in an apartment, that's a moot point.

Those in the market for a full-range loudspeaker for use in rooms under 600ft² in area should make the Audio Artistry Dvorak a top priority on their audition lists, regardless of budget. Those planning on spending much more may be pleasantly surprised, and those with tighter budgets are likely to find themselves justifying "creative financing" alternatives in order to take these babies home. Happy listening!

—**Shannon Dickson**

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