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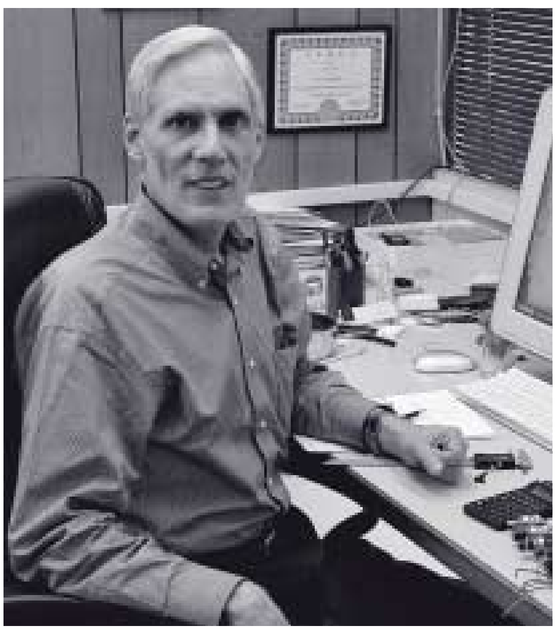
An Interview with Jim THIEL

by Richard Hardesty

I've known Jim Thiel (cofounder and chief designer) and Kathy Gornik (co-owner and company president) since the late '70s when I became a dealer for their speaker products. I sold, installed and repaired Thiel speakers for many years and am quite familiar with the quality of their construction and the people who make them.

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



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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

Ideally we want a driver diaphragm to be infinitely stiff and light enough so that we can maintain reasonably high efficiencies.

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

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

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

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

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

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

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

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

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

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

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

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

Exactly!

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

I guess so. But it's a pleasant surprise that it turns out there are people out there that can hear the difference—maybe not "most" people, maybe not a large enough group to allow me to buy a great mansion. But there are people who appreciate it a lot. We get letters all the time from customers telling us they've gotten so much enjoyment from our speakers, even the ones

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

they bought 15 years ago, and thanking us for doing what we did. And that's pretty neat!

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

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

Like a lot of things in loudspeaker design, solving one problem can create other problems. In this case, we found that you can't simply mount a tweeter in the center of a woofer because that

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

As you said, many of our coaxial drivers are more or less conventional in that there is a self-contained tweeter with its own motor system that is mounted within the self-contained midrange with its own motor system. But we do make some coaxial drive units that share the same motor system and use a mechanical crossover. There's only one magnet system and one voice coil, but that voice coil is connected to a mechanical crossover system that allows only the lower frequencies to reach the midrange diaphragm, whereas all of the frequencies go to the tweeter diaphragm, including the very high frequencies.

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

That's exactly right. The coupling suspension, as we call it,

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

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

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

Really? I'd never encountered that before. I've seen "wizzer" cones used in attempts to extend high frequency response of midranges but I've never seen one with a mechanically decoupling crossover… I looked at that and thought, Gee, if he tells me how this works he'll probably have to kill me! This has got to be secret.

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

properly in practice was not so easy.

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

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

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

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

Stressing what we just talked about, ideally the whole cabinet needs to be so rigid that it will not vibrate. You have these drivers that are producing sound by vibrating in and out and sound is radiated from the front of the diaphragm and that's what we hear. But sound is also radiated from the rear of the diaphragm and goes into the speaker cabinet. The job of the enclosure is

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to completely contain that energy so it doesn't get out into the room to distort the sound that you're hearing—and that becomes very difficult.

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

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

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

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

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

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

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

the response that it needs to provide is different depending on which drivers it's working with.

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

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

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

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

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

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

For lower frequencies I need to measure outside. It's tricky to get such measurements but the frequency response measurements are by far the most useful. There are some caveats about that. What would look good on paper might not sound like a good frequency response measurement.

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

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

measure flat within ±2dB and not sound good at all, and another might not measure any better in terms of frequency response but sound quite good.

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

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

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

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

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

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

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

Tell us about your subwoofers.

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

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

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

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

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

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

And you do that with that "smart" crossover.

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

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

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

That sounds like a very complicated device.

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

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

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

Thank you Jim for an informative and enlightening interview!

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

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