DAVID RICH REVIEWS A TOTALLY NEW KIND OF AV EQUIPMENT.

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ISSUE NO. 29
Display until arrival of Issue No. 30.

The Audio Critic

"Hip Boots,"

our classic column that relentlessly waded through the mire of misinformation in the audio press, comes to a reluctant but inevitable end.

Also in this issue:

A slew of unusually thorough loudspeaker reviews, by Don Keele, Tom Nousaine, and Glenn Strauss.

Reviews of AV electronics, power amplifiers, and assorted other electronic components and accessories.

Plus our standard features, columns, letters to the Editor, CD/SACD/DVD reviews, etc.

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ISSUE NO.29 • SUMMER/FALL 2003

Still in Transition.

This is the first magazine I've ever edited completely, and the first issue of ***The Audio Critic*** that Peter Aczel has not. He's neither gone nor going, just taking a step back and letting someone else—me—handle the daily work.

In the last issue, Peter introduced me as the former Technical Editor of *Audio,* but I feel some further introduction is in order.

I've been writing about audio since my beard was black, stereo was new, and everything was analog. I appreciate good sound, maintain a healthy skepticism about the "scientific" claims of manufacturers and designers, and realize that audio components can sound good even when those claims do not make sense—and sound bad even when they do.

My writings have appeared in many magazines, several languages (including Portuguese and Flemish), and under a few pen names. (At one time, I was writing for *Audio, Stereo Review,* and *High Fidelity,* which the magazines didn't mind as long as I used a separate name for each; an editor once called me "three of the best-known hi-fi writers in America.") I've also been an editor at *Popular Mechanics, Popular Electronics,* and *Video,* as well as *Audio.* It's my editing that counts, here, and you can judge that for yourselves.

Of the other transitions Peter Aczel mentioned in the last issue, the partnership with The CM Group stopped progressing, and is gone. Progress has been made in getting us caught up with our four-time-a-year schedule; we're far from there, as yet, but we're continuing to move in that direction. This issue came a little faster than the last; the next should come faster still.

Ivan Berger

After Ivan Berger had written the above, further unforeseen delays took place. What's more, and worse, we lost our second-class mailing permit, "due to nonuse" (too true, alas). The consequences remain to be assessed. Will we continue to publish? You bet. Adversity just makes us more stubborn. —Ed. (RA.)

The Audio Critic

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Some of our readers still don't understand what kind of letter is likely to be published in this column. Zen hint to the intuitive: not the kind that begins with "I have a Schmigehgie QX-200 amplifier-in your opinion is it. . ." Please address all editorial correspondence to the Editor, The Audio Critic, P.O. Box 978, Quakertown, PA 18951 0978.

The Audio Critic:

I hope Issue No. 28 was not the last issue of *The Audio Critic,* since it shines by an absence of the typical technical nonsense that I find in all the other audio magazines. On top of that the tone of presentation makes it so much more readable than often in the past.

In the introduction to "Speakers" (page 15), you point out that good speaker design is the sum of many, many aspects that were properly dealt with. I agree wholeheartedly, and also with your example of the Waveform Mach 17 speaker system. Yet this speaker, and all the ones that Floyd Toole referred to in the feature article, suffer from being caught in the box paradigm. The ultimate performance and accuracy of reproduction that can be achieved within this paradigm are limited, and the best of Toole's examples have reached that plateau.

There are two fundamental problems with box speakers: (1) selective reradiation of the sound energy inside the box through the cone and walls and (2) a power response, or directivity index, that changes at least 10 dB between low and high frequencies. The reradiation problem has been addressed to varying degrees of success by different designers. Constant power response, though, requires drivers and box features that decrease in size as frequency increases, to maintain wide and uniform polar response beyond what the speakers in Toole's article achieve. The result of failing to deal with these two problems is the typical, generic box-loudspeaker sound that is immediately recognized in comparison to live, unamplified sounds.

Loudspeakers end up in rooms. The off-axis radiation therefore matters, as Toole's findings clearly point out, but 10 dB variation in power response is too much and limits this design approach. The improvement beyond it is via omnidirectional box speakers or full-range, open-baffle dipole speakers. Some planar electrostatic or magnetic designs show the potential of this approach, but ultimately they are limited by being acoustically too large at higher frequencies, yet having insufficient volume displacement for low-frequency reproduction at near realistic levels. These problems can be overcome with conventional dynamic drivers on open baffles. As it turns out, such speakers are significantly less sensitive to the room both below and above 500 Hz.

The "preservation of the art" problem, or the "circle of confusion," can only be resolved by using unamplified sound as a reference and not other loudspeakers. This will also point out the need to reduce nonlinear distortion and stored energy, which are at least equal in importance to the different steady-state frequency responses.

Siegfried Linkwitz
Linkwitz Lab
Corte Madera, CA

Siegfried Linkwitz is one of the most distinguished practitioners in audio—what audiophile hasn't heard of the Linkwitz-Riley crossover? He was a Hewlett-Packard scientist before he started designing loudspeaker systems for Audio Artistry and Linkwitz Lab, all of which are based on the dipole principle. Thus the above letter is motivated by a designer's agenda, but that doesn't make it less valid. The arguments in favor of the dipole approach are powerful and not to be ignored. We wish we could test one of the Linkwitz-designed loudspeakers—how about it, Siegfried? We listed you as one of the White Hats (good guys of audio) in Issue No. 24, but that was based mainly on your engineering papers and spoken commentary, not specific products. It's time for some hands-on. As for your favorable comments on our publication, they couldn't come from a more authoritative source and are therefore especially welcome.

—Ed.

The Audio Critic:

In response to Issue No. 28, "Science in the Service of Art"—is Floyd E. Toole colorblind? First picture: a portrait painted under a light with 3 dB too much red versus when viewed under natural, neutral light. I've been painting for over 40 years and I never knew light could be measured in dB.

As for the rest of your magazine, you've done better. As far as your plan to retire to a primarily supervisory position is concerned, just retire and let Ivan Berger take over.

Your "Hip Boots" column is sorely missed because it keeps the crazy audio drivel in check. *Don't give that up!* Tom Nousaine's "Urban Audio Legends" comes close but does not have the "bite" of "Hip Boots."

Maron Horonzak
Stoutsville, MO

Marrone, *Maron! You never knew that light could be measured in dB? Well, it seems there are lots of things you never knew, and this is one of them. A dB number can be just an expression of a ratio, e.g., 20 dB is a ratio of 10 to 1, 10 dB is a ratio of 3.16 to 1,3 dB is a ratio of l.41 to 1. Thus 3 dB too much red means 1.41 times as much red as there should be—41 % too much. Your assumption that it's Floyd Toole, Ph.D., who doesn't know what he is talking about, rather than you, reveals a lot about you.*

Now, about Issue No. 28 not being as good as some others, you may be right. When there are three or more of anything, one will be the best, one will be the least good, and the other(s) will be in between. That doesn't mean, however, that they aren't all good. As for my total retirement and letting Ivan Berger take over, it's a staggeringly simplistic suggestion innocent of all business/financial/professional/personal considerations. Didn't it occur to you that it just might be more complicated than that? lastly, "Hip Boots" is back in this issue but, as explained there, not as a continuing feature. Thanks for all your concerns.

—Ed.

The Audio Critic:

I was truly delighted to find Issue No. 28 in my mailbox yesterday. It is so good to see that someone is still out there battling the fakes and frauds. Peter, you are my hero. *[Mutual admiration society! See below.—Ed.]*

It is most pleasing to see the excellent authors you have corralled and the fine articles that you have published. In your editorial you claim to be getting old and tired. But cheer up. What you are doing with *The Audio Critic* is such excellent work that it must go on.

I have retired from the audio field after many years and am now, 10 years into retirement, simply relaxing with my music and other hobbies. These are gardening, astronomy, and mineral collecting. Still, I think about audio matters very often and still do a bit of consulting in room acoustics and audio systems.

I have taken the liberty of sending you a couple of photos of my listening room as it is now and has been for 22 years. I am still pleased with it and find no reason to change anything. It is now the music that counts for me.

Very best regards and best wishes for future success.

Sincerely,
Dick Greiner
Madison, WI

Dr. R. A. Greiner is Emeritus Professor of Electrical and Computer Engineering, University of Wisconsin, and one of my heroes, as our regular readers know. For quite a few decades before his retirement he embodied the academic community's most authoritative, and at the same time most genial, voice on the subject of audio. Talk about "battling the fakes and frauds"—he was at all times in the font lines, patiently refuting charlatanry with irrefutable science. My admiration for him is unlimited, hence his frequent presence in this column. We may not have anything near the circulation of Stereophile, *but could they ever, in a million years, have elicited a letter like the above from Dick Greiner?*

As for your music system and listening room, Dick, should I be surprised that you are not looking for a change? What, only eight monstrous woofers? Only 24 visible smaller drivers? Only a dozen electronic units? I have never seen a 1980 setup like yours, and very, very few 21st century rigs like it. It really amuses me when you say that only the music matters; it's like a Rolls Royce owner saying that, well, it's basic transportation. May you listen to that music in good health and spirits for many years to come—and thank you for your compliments.

—Ed.

The Audio Critic:

Hello Peter, I have come to praise you, not to bury you!

Item One: I received the latest issue of *The Audio Critic* (No. 28) and immediately went to die "From the Editor" column. Your explanation as to the reason for the disintegration of your relationship with The CM Group caught my attention. Wanting to hear the other side of the story, I placed a call to The CM Group publisher, Greg Keilty . . .

. . . Greg had not read your explanation as to what happened between you and The CM Group, so I read your explanation to him (verbatim). Was I surprised at his response! He agreed with you completely! To be completely honest (which is a much better form of honesty than partially honest!), I was expecting at least a minor disagreement from Greg regarding your explanation. There wasn't. Not only did he agree with your explanation but spoke very highly of you! Son of a gun! You get an A+ for editorial integrity and my apology for doubting your word. It's somewhat humbling to admit I was wrong, but it would be a mortal sin not to admit so and apologize.

Item Two: I received the latest issue of *Invention & Technology* magazine (Fall 2002). Within this issue of the magazine was an article (the cover story) titled "The Tube Is Dead, Long Live the Tube," written by Mark Wolverton. No need to tell you that I couldn't wait to read Mr. Wolverton's article. I was expecting more of the idiotic subjective audio-cult gibberish printed as fact by mainstream publications *{Wall Street Journal, Business Week, Fortune,* to name a few). Fortunately, this time, the cultists were shown as believing (?) and propagating myths based only on their emotional or financial involvement with tube equipment—something you have been preaching for quite a while. Much to my pleasant surprise, you and David Rich were quoted regarding the

(continued on page 44)

By PETER ACZEL, Editor

Our Last Hip Boots Column

How come? Because "wading through the mire of misinformation in the audio press" (our former subtitle) is no longer meaningful when nearly the entire audio press is dedicated to misinformation.

Before we stopped running our "Hip Boots" column three issues ago, its subject was almost invariably the ignorant and/or irresponsible subjectivity of certain audio reviewers, more often than not Bob Harley (ignorant) or John Atkinson (irresponsible) or Harry Pearson (ignorant and irresponsible). Occasionally we addressed purely technical errors in various publications, sometimes even the mass media, that a good fact checker could have corrected, but most of the time our target was the absence of accountability in one or the other of the same three or four audio magazines. That's where the mire lay that only hip boots could wade through.

Lately the ground has shifted—or, rather, it has expanded, spread out, in a totally engulfing mode. We have reached the point where virtually the entire audio press is in tacit denial of the realities of electrical engineering and electroacoustics. All debate on the subject has ceased. The false assumptions we used to attack have become the self-evident givens of the audio journalists. Fiction is now accepted fact, mindless misinformation is unquestioned mainstream. So—what's the point of singling out individual examples of this sad state of affairs? Whatever nonsense reviewer X writes is echoed just as unthinkingly and self-assuredly by reviewer Y and reviewer Z. Why "hip boot" X but not Y or Z?

Do you think I'm overreacting or exaggerating? Then tell me which equipment reviewer refrains from ascribing a personality to amplifiers, preamplifiers, and CD players. They all do it, except David Ranada, technical editor of *Sound&Vision,* the one magazine that is at least a partial exception to the rule—but their other reviewers are not as careful and tend to fall into the trap of characterizing the sound of electronic equipment.

For our newer readers I should perhaps point out all over again the pathetic fallacy of talking about the soundstaging, or front-to-back depth, or open/closed quality, or graininess, or any other sonic characteristic of purely electronic signal paths that are less than, say, 20 years old. What the human ear can differentiate are frequency response, level, noise, and to a lesser extent distortion. That's all. Since all modern audio components, from a $15,000 rip-off amplifier to a $69 portable CD player, have flat frequency response, negligible noise, and negligible distortion, their sound has no signature, no personality. Any two of them—two amplifiers, two preamps, two CD players, etc.—will sound exactly the same, as long as their levels are matched within ±0.15 dB. I solemnly guarantee it. There has not been a single properly conducted listening test—double blind, at matched levels—to contradict that statement. This will surprise only some of the aforementioned newer readers and elicit a chorus of denial from the more obstinate of the high-end reviewers, but it is an ironclad truth. Think about it. There is no such thing as an effect without a cause, and what could cause a sonic difference except a skewed frequency response, a high noise floor, or unusually high distortion? What you are told in *Stereophile, The Absolute Sound,* and other such publications is arbitrary effect without an explainable cause—"Hip Boots" material over and over again.

I'll grant maybe a rare exception to the above in the case of the most eccentric "retro" vacuum-tube designs, which depart so radically from the flat/low-noise/low-distortion model that, for all I know (and I don't care), they sound different. The gullible are welcome to these electronic abortions. I'll also grant that matching levels within ±0.15 dB (preferably within ±0.1 dB) is a fussy, sweaty, boring process, requiring some instrumentation, and for all those reasons not done when it should be. That is unquestionably the main nonideological reason for all the stonewalling denials of the soundalike outcomes. (Larry Klein, former technical editor *of Stereo Review, Sound & Visions* predecessor, once suggested a delightfully ironic solution to this problem. He said you don't need any instrumentation to match levels within ±0.1 dB; all you need to do is fuss with the volume controls until A and B sound exactly alike, at which point the levels will be perfectly matched. Bingo! I love it!)

Let us also address the opposite end of the spectrum, where there are always large differences in sound—loudspeakers. Every loudspeaker ever made is at least slightly different in frequency response from every other and therefore necessarily sounds different. Unfortunately, this creates another likely "Hip Boots" situation. The various subjective equipment reviewers have no clue as to how to relate the measured performance of a loudspeaker—if indeed they have measured it—to its sound. Let us say it has rapidly falling low-frequency response below 60 Hz. In that case they would probably praise its superior bass. Or it has almost dead-flat high-frequency response over a large angle. In that case they would complain about its attenuated treble. If it has a huge suckout in the crossover region at, say, 1.8 kHz, they would praise the highly accurate upper midrange. And so on. It isn't just one or two reviewers that do this. I look at the subjective high-end magazines and find absolutely no correlation between measured performance and listening appraisal. Not that more than one or two of them do any measuring at all, but I do and I can't find a single loudspeaker reviewer whose perceptions agree with mine and track my measurements. (Tom Nousaine of *Sound & Vision* is an exception, but he doesn't count because he also writes for *The Audio Critic.)* Since none of them do an orderly, logical, disciplined job—like Don Keele or David Rich or me in this publication—what's the point of "hip booting" one or two of them?

Rethinking the Comparative A/B Listening Test

My conclusions from all this actually go beyond the futility of continuing "Hip Boots." I am beginning to think that all comparative (A/B) listening tests have become unnecessary. What? How can an audio equipment reviewer possibly be saying this? Bear with me for a moment. Since all modern electronic signal paths sound the same, why go through the motions of A/B-ing them? I can guarantee that the results will be the same over and over again, and therefore the exercise is a waste of time. (Don't misunderstand me. I'm not suggesting that we stop listening to and enjoying music through the various components. I'm talking about A/B-ing.) You can try it, as I have, with a multikilobuck high-end amplifier (A) and a dirt-cheap Japanese mass-market receiver (B). You'll see.

But that's not all. I'll go further. Why A/B loudspeakers? The flatter they are in frequency response and the lower they are in distortion, the more nearly they will approach total neutrality, total transparency, which is both the goal and the reference point. Conversely, the more they deviate from flat and distortion-free response, the more they will deviate from neutrality/transparency. These relationships, as I have found out over the years, are linear—the approximation of the ideal sound is exactly proportional to the degree of perfection obtained from the measurements. In other words, there are no surprises in the listening tests—in which case why bother with them? The only reason to do so is that our measurements are incomplete; we would need the 72 different measurement points in a 4π space that Floyd Toole uses at Harman International to be sure that we have characterized each speaker completely. If we had the laboratory facilities to do that, I wouldn't A/B test anymore (although Floyd still does). We have reached the point in audio where the laboratory instruments know it all and tell it all, much as the golden-ear boys hate to admit it.

All of the above considerations are made more complicated by surround sound, which has its own rules. It is not easy to understand that the audible differences between Dolby Pro Logic (I and II), Dolby Digital, DTS, Home THX Cinema, 5.1, 6.1, 7.1, etc., etc., are not a matter of signal paths but of algorithms. The differences are determined mathematically, not acoustically. (There are also differences in bit rate between Dolby Digital and DTS, but none that I consider to have audible significance.) Listening tests, therefore, are quite limited when it comes to sorting out the inherent audible characteristics of each configuration, because basically the sound is determined before it reaches the amplifier/speaker stage. It could actually be better studied from a block diagram. Again, remember that I'm talking about comparative listening, not musical enjoyment.

In general, the paradigms have shifted, journalistically, electroacoustically, psychoacoustically, every which way. You can't take the old perspectives for granted. We are well into the 21st century. Or perhaps I should say, discomforting as it is, we aren't in Kansas anymore. TAC

By Ivan Berger, Guest Editor
D. B. Keele Jr., Contributing Editor
Tom Nousaine, Contributing Editor
Glenn O. Strauss, Contributing Editor

Five Loudspeakers (One a Time-Honored Exotic) and a Headphone

2-Way Audio/Video Minimonitor Loudspeaker

Definitive Technology StudioMonitor 450

Definitive Technology, 11433 Cronridge Drive, Owens Mills, MD 21117. Voice: (800) 228-7148. Fax: (410) 363-9998. E-mail: info@definitivetech.com. Web: www.definitivetech.com. StudioMonitor 450 shielded audio/video minimonitor loudspeaker. $329.00 each ($658.00 the pair). Tested samples on loan from the manufacturer.

Introduction

A 10-inch, side-mounted woofer in a full-range speaker this small? Not quite. It's something Definitive Technologies calls a "planar-technology pressure-driven subwoofer"—in other words, a passive radiator (sometimes called a PR, drone cone, auxiliary bass radiator, or flapping baffle). Typically, passive radiators substitute for vents, or ports, in small speakers designed to deliver substantial bass output. That's a good description of the StudioMonitor 450, a member of Definitive Technologies' "Monitor Series" of modestly priced home-theater loudspeakers.

Using a ported cabinet instead of a closed box extends a speaker's low-frequency response and reduces its distortion. It does this by coupling an acoustic resonant system (the enclosure and a port—usually a tube—that vents its output into the room) to the rear of the speaker's diaphragm. This sets up an acoustic resonance between the mass of air moving in the port and the stiffness of the air in the enclosure. By matching the enclosure and port sizes to the characteristics of the driver, this resonator is typically tuned to a frequency near the lowest frequency the system is intended to reproduce, and radiates low frequencies over a range of roughly two-thirds of an octave around its resonance. If the system is properly designed, far more sound comes from the port (within its operating range) than from the driver. Because the ported enclosure's acoustic resonant system is typically more linear than the mechanical resonant system of the speaker, its distortion is lower. Below its resonance, unfortunately, the port's output is essentially out of phase with the loudspeaker's, which makes the system's bass output roll off much faster than that of an equivalent closed-box system.

Manufacturer's Specifications

Type: 2-way, vented-box, shielded audio/video monitor

Drivers: 6½" cast-frame bass-midrange woofer, 10" side-mounted passive radiator, 1" aluminum-dome tweeter

Crossover Frequency: Not stated

Rated Frequency Response: 24 Hz to 30 kHz, tolerance not stated

Rated Sensitivity: 90 dB at 1 meter, 2.83 V rms applied

Rated Impedance: 4 to 8 ohms, nominal

Recommended Amplifier Power: 20 to 180 watts

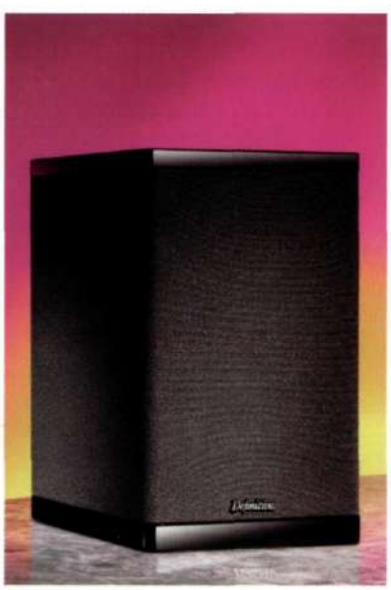
Dimensions: 13" (height) x 8⅜" (width) x 11⁷⁄₁₆" (depth) = 330 mm x 213 mm x 291 mm

Weight: 18 lbs. (8.2 kg) each

Finish: Piano-gloss black, white, or golden cherry

There is, however, a catch to all this: At high volume levels, the air in the vent can move fast enough to generate significant turbulence, which causes extraneous noise and limits the port's output. This turbulence can be tamed by increasing the port's area, but that calls for lengthening the port to increase the air mass within it. Otherwise, the box resonance, and hence the shape of the speaker's response curve, will change. For small boxes that are tuned to low frequencies and designed to radiate a lot of acoustic power, enlarging the port's mouth would call for very long port tubes that take up a lot of space in the box; sometimes, tubes that are long enough won't fit!

Using a passive radiator sidesteps these problems. Typically, a passive radiator is a speaker (frame, cone, surround, and sometimes spider) without a magnet and voice coil. Here, the ported-box resonance is a function of the mass of the passive radiator and the compliance of the air trapped in the enclosure. Because it is shallow, the radiator can be made large enough to avoid turbulence while taking up hardly any space within the cabinet. And because the radiator's mass is independent of its area, any size will do as long as its air-moving capability (area times stroke) is sufficient. A properly designed passive radiator requires roughly two to four times the air-moving capability (1.5 to 2 times the diameter) of its companion driver and must have a self, or free-air, mechanical resonance at least an octave below box resonance.



In Definitive Technology's compact, two-way StudioMonitor 450, the companion driver to the 10-inch passive radiator is a 6½-inch cone woofer/midrange, used with a 1-inch aluminum-dome tweeter. Both active drivers are mounted on the front of the cabinet, with the tweeter on top and offset about an inch to one side. The speakers are provided in mirror-image pairs, with black, white, or golden-cherry piano-gloss finishes on the top and bottom. The front, sides, and rear are covered in a wrap-around grille cloth, held in place by the removable top and bottom pieces. Connection is through a single pair of gold-plated multiway binding posts, spaced for double banana plugs, on the bottom rear of the cabinet. Cabinet construction is quite heavy-duty for a speaker system of this size and price: The medium-density fiberboard (MDF) front panel is a full inch thick, and the remaining panels are ¾-inch MDF. The cabinet is well braced and quite solid.

The magnetically shielded, 1-inch aluminum-dome tweeter is essentially the same as that used in Definitive Technology's top-of-the-line systems. The 6½-inch bass/midrange driver, also magnetically shielded, has a cast basket. The 10-inch passive radiator is simply a rigid circular plate with an attached surround.

The SM 450's crossover is wired on a small PC board mounted near the speaker's input cup and can be reached by removing the cup. The crossover, a second-order design, has an iron-core inductor connecting the woofer/midrange, an air-core inductor in the tweeter circuit, four power resistors, and three capacitors. That's one more resistor and capacitor than usual; the extra components probably act as an impedance-compensating network.

Measurements

As in my previous reviews, I used two different test techniques to measure frequency responses. I used nearfield measurements to assess low-frequency response, and measured response at middle to high frequencies with windowed in-room tests (my test microphone was centered between the tweeter and woofer/midrange axes, 1 meter away). The test signal for these measurements was the usual 2.83 V rms, and the curves were subjected to one-tenth-octave smoothing.

The on-axis response of the Studio Monitor 450 is shown in Fig. 1, including smoothed and unsmoothed responses above 10 kHz. Only the response with the grille on is shown, because the grille cloth is not designed to be removed; fortunately, the grille had essentially no effect on the SM 450's response except for very small deviations of less than ±0.5 dB between 8 and 12 kHz. The smoothed curve is quite well behaved and fits a tight (2.5-dB) window from about 95 Hz to 12 kHz. At higher frequencies, the unsmoothed curve exhibits a sharp dip of about 10 dB at 12.9 kHz followed by a less energetic peak of about 4 dB at 14.5 kHz, both presumably caused by a resonance in the tweeter's metal dome. At low frequencies, the system rolls off slowly, reaching -3 dB at 83 Hz, -6 dB at 63 Hz, and -9 dB at about 50 Hz (which is near the SM 450's vented-box resonance). Below 50 Hz, the system rolls off rapidly, about 24 dB per octave, as is common with vented-box systems. However, this curve was measured in free space,

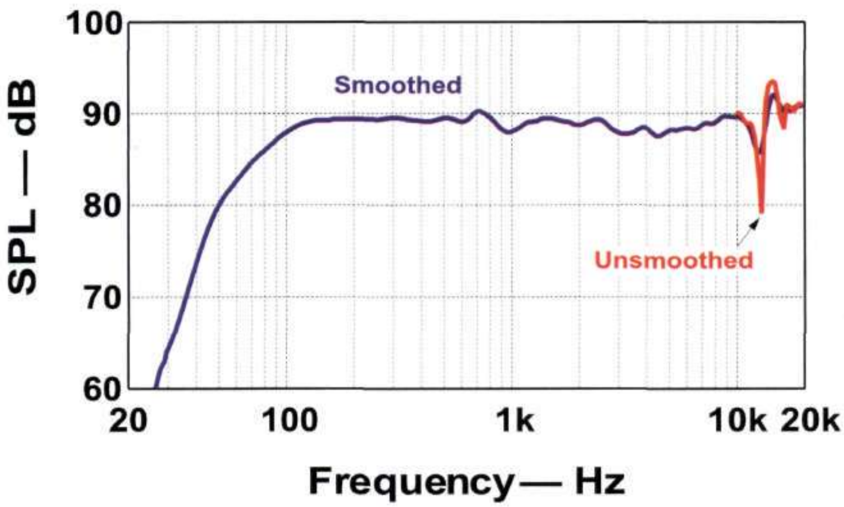


Fig. 1: On-axis frequency response.

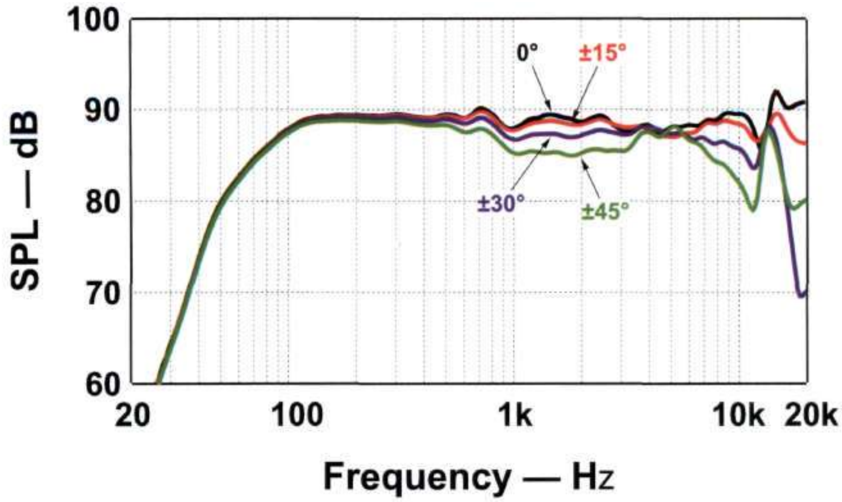


Fig. 2: Horizontal off-axis frequency response.

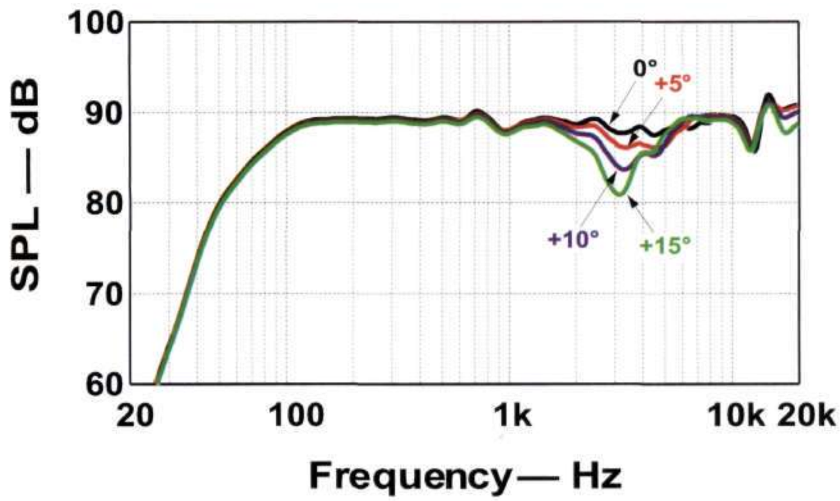


Fig. 3a: Frequency response above axis.

without reflecting surfaces to augment the bass; in a room, reflections from the walls would enhance the bass considerably. Averaged between 250 Hz and 4 kHz, the SM 450's sensitivity was high (89 dB), just 1 dB less than Definitive Technology specifies. The left and right speakers matched within + 1 dB, with most of the difference occurring in the tweeters' range.

The SM 450's horizontal and vertical off-axis frequency responses are shown in Fig. 2 and 3. Fig. 2 shows the horizontal off-axis curves, in 15° increments out to ±45°. Between 1 and 3 kHz the response shelves downward, the dip worsening as the off-axis angle is increased. There are also significant high-frequency aberrations above 8 kHz at extreme off-axis angles. These aberrations include a dip above 10 kHz, followed by a peak at about 13 kHz.

That high-frequency dip and peak are also seen in the responses measured above and below the tweeter's axis. The above-axis curves (Fig. 3a) are quite well-behaved, except for a dip in the 3 kHz crossover region that deepens progressively as the listening angle increases, and the previously mentioned aberrations above 10 kHz. Response below axis (Fig. 3b) is significantly smoother. Between 3.5 kHz and 7 kHz the output below axis slightly exceeds the on-axis output, which indicates that the SM 450's response is smoothest a bit below axis. This implies that these speakers should be aimed above ear level, or possibly be mounted upside down to provide the smoothest response for seated or standing listeners. Luckily, the cabinet bottoms are finished like the tops, although there are four small bumps that serve as feet. Unfortunately, the logos on front of the speakers are upside down when the speakers are inverted.

The input impedance magnitude of the StudioMonitor 450 (Fig. 4a) drops to a low of 3.2 ohms in the lower midrange (at about 200 Hz) and reaches a high of about 14 ohms slightly below crossover, at 1.6 kHz. The two impedance peaks that mark the 450 as a vented box are clearly evident; the impedance minimum (3.7 ohms at about 55 Hz) shows where the box is tuned. The impedance phase (Fig. 4b) is well behaved and varies only moderately, about ±38°. The SM 450 should be an easy load for any competent power amplifier or home-theater receiver.

To measure the distortion of the 450 (Figs. 5a and 5b), I used some software I recently wrote that works in conjunction with Igor Pro 4.0, a graphics and data-analysis program (available for Mac and PC from www.wavemetrics.com) and an external audio interface with 24-bit A/D and D/A converters, the Sound Devices USBPre (www.sounddevices.com). In this setup, test signals generated by Igor are fed through the USBPre and my amplifier to the 450s, while signals from my test microphone are fed to the computer through the USBPre, then analyzed and plotted on graphs by Igor.

Fig. 5a shows the sine-wave harmonic distortion of the 450, evaluated from 40 to 500 Hz at frequencies $^1/_{12}$ octave apart. The distortion was evaluated at each frequency by applying a sine wave to the system for one half second and then evaluating the harmonic distortion of the system's output, measuring the total energy of the 2nd through 5th harmonics by using FFT (Fast Fourier Transform) to compute the frequency spectrum of that output. (Results are expressed as a percentage of the fundamental's signal level, not as a percentage of the total output. Note that this calculation method allows distortion levels above 100% if the energy of the harmonics is greater than the energy of the fundamental.) The harmonic distortion at each frequency was evaluated at three different power levels, 6 dB apart. (The

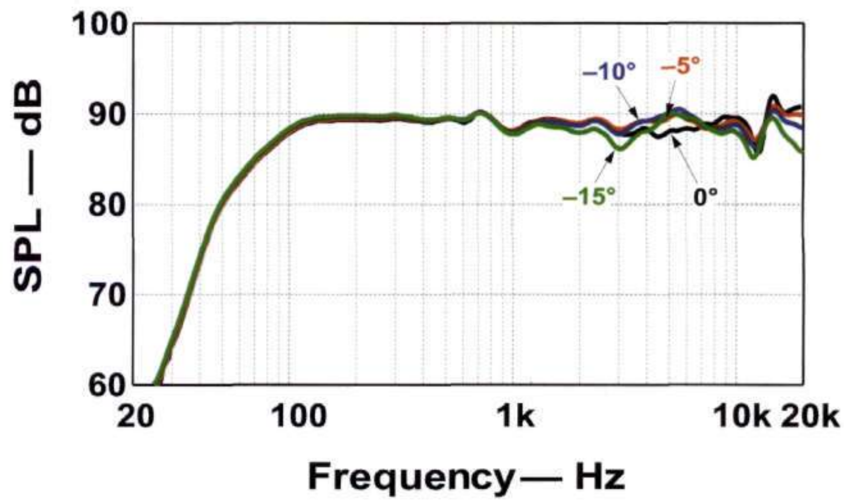


Fig. 3b: Frequency response below axis.

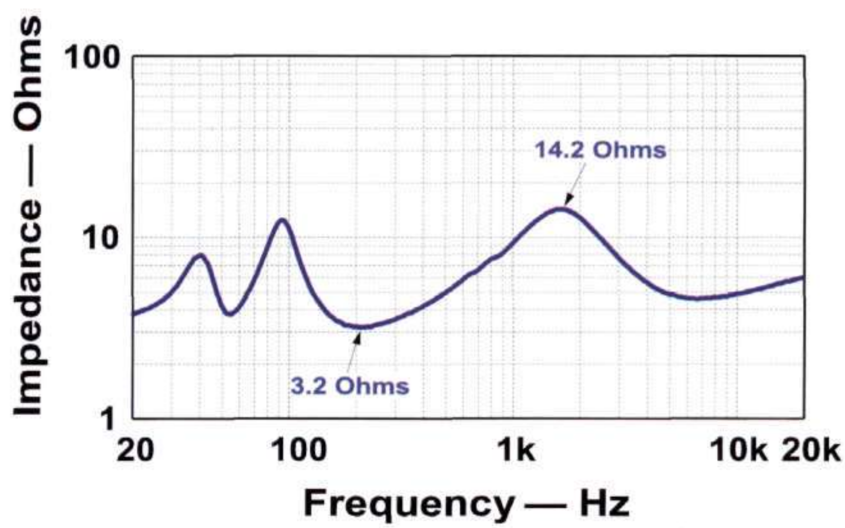


Fig. 4a: Impedance magnitude.

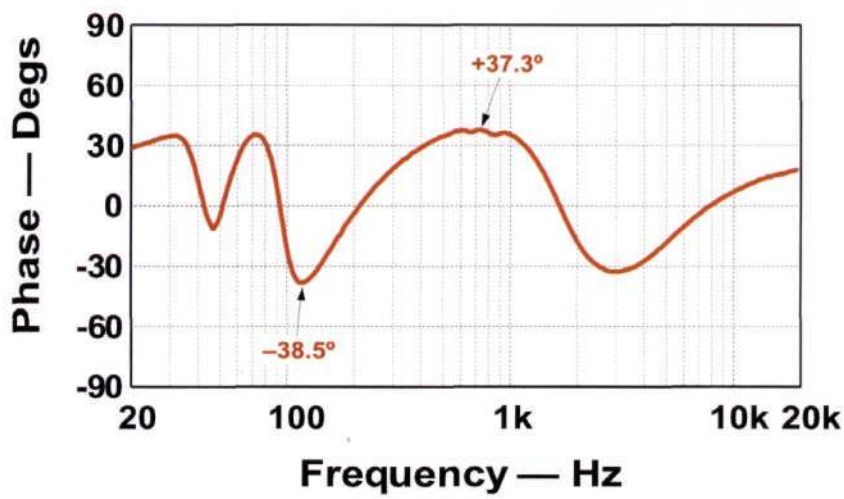


Fig. 4b: Impedance phase.

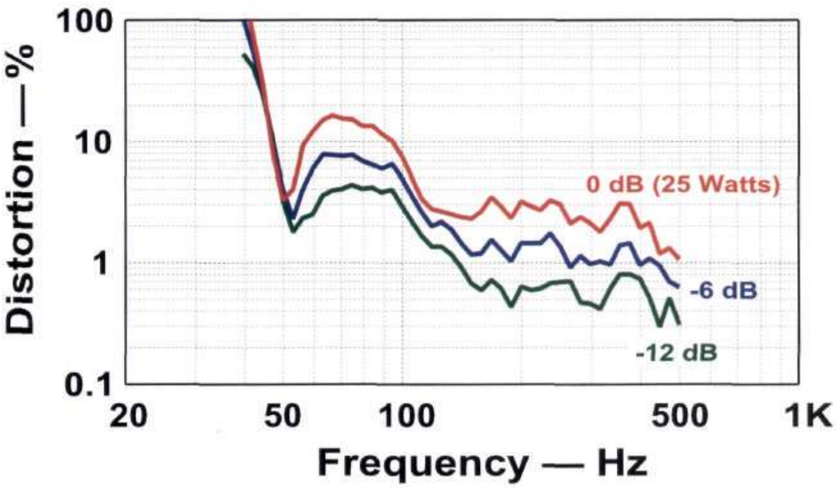


Fig. 5a: Harmonic distortion versus frequency and power level.

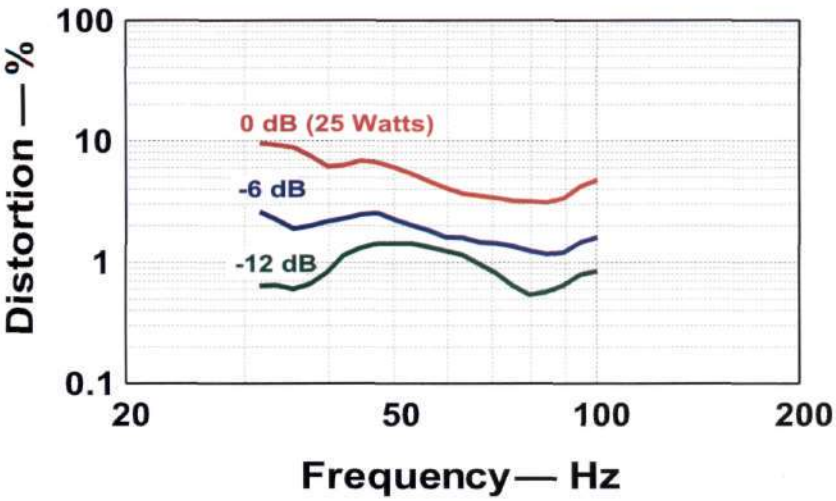


Fig. 5b: Intermodulation distortion versus frequency and power level.

0 dB level was 2.5 V rms, or 25 watts assuming a 4-ohm impedance.) Measurements were made with the speaker on its side on the ground plane, passive radiator facing up, and the test microphone at the ground plane, on the woofer/midrange driver's axis and 0.25 meters away.

For frequencies above 125 Hz, the harmonic distortion percentage stays roughly constant and generally doubles when the input power does. Below 125 Hz, the distortion reaches a maximum at about 70 Hz, falls to a minimum between 50 and 55 Hz, and then rises rapidly at lower frequencies. The dip in the vicinity of 50 Hz coincides with the system's vented-box tuning frequency, where the passive radiator is producing most of the sound. (As you can see from the slight shift in this dip when the power level changes, box tuning varies slightly with the test conditions; this is why the impedance measurement, above, indicates 55 Hz as the tuning frequency.) At the highest power level, 25 watts, the maximum distortion is a moderate 16% or so, occurring at 70 Hz. At the power levels I used for this test, the distortion did not become irritating until the test frequency dropped below 45 Hz.

The SM 450 woofer's intermodulation distortion (IM) was measured with the same power levels and test conditions as in the harmonic distortion test but over a slightly different range of frequencies. For this test, I applied two tones of equal level, one fixed at 440 Hz, the other varying from 31.6 to 100 Hz in half-octave steps. The dual-tone test signals were applied to the speaker for one half second each. The test results, expressed as a percentage of the energy of the two original test tones, represent the total energy of three intermodulation sidebands above and three below the higher test frequency. The IM (Fig. 5b) varies slightly over the tested frequency range and increases as the power level increases. At the highest test level (25 watts) the IM rises to roughly 10% at the lowest test modulating frequency. While 10% harmonic distortion is not annoying, 10% IM distortion is. At power levels of -6 dB (5 watts) and less, the IM remains below 3%.

Use and Listening Tests

When I first unpacked the StudioMonitor 450s, I was quite impressed with their overall appearance, especially the cabinets' piano-black top and bottom panels. At first, I could not figure out how to get the grille cloth off so I could see the drivers, but I soon determined that the top and bottom panels could be removed, as they are attached to the cabinet with four pegs that engage holes in the panels. When a panel is removed, it uncovers the grille cloth, which is tightened around the cabinet with a captive drawstring. The grille wraps completely around the cabinet and has a cutout at the rear for the input-terminal cup.

When uncovered, the speakers and cabinet had a meticulous, no-nonsense look that showed careful craftsmanship and attention to detail. Under the grille cloth, the enclosure was finished in an attractive satin black. The SM 450s are provided with wall-mounting brackets that screw into routed-out holes on the rear panel—a nice touch.

The large passive radiator essentially takes up one whole side of the cabinet; in an enclosure this size, it looks like a monster woofer. The radiator is inset " to protect it from damage. When energized by high-level sine waves, the speaker sounded quite clean down to 40 Hz, but distortion was audibly significant at lower frequencies. At the box tuning frequency, the woofer's motion almost ceased and the passive radiator's excursion became quite large. The deep null in the woofer's excursion showed that the box and the passive radiator work extremely well.

At and near the system's tuning frequency, maximum clean excursion was about 0.3" peak-to-peak for the woofer and a healthy 0.4" peak-to-peak for the passive radiator. The effective radiating diameter of the passive radiator is about 8.35" and that of the woofer about 5". This makes the drone cone's radiating area approximately 2.7 times that of the woofer—and with its higher excursion capability, it can move roughly 3 to 3½ times as much air as the woofer. As I said above, this is good design practice for a passive-radiator system.

For my listening, I placed the systems on 24" stands (which raised the tweeter to about 34½" above the floor) about 7 feet apart and well away from room's side walls. I drove them with my Crown Macro Reference power amplifier and Krell KRC preamp. The SM 450s are smooth-sounding speakers, and their sensitivity is quite high, especially as compared to my reference B&W Matrix 801 Series 3 systems. In A/B comparison tests, I had to attenuate the input to the Definitive Technology speakers by about 4 to 4.5 dB to match their levels to the B&Ws'. The LED level monitors on my power amplifier showed that the amp was working noticeably less hard when driving the SM 450s. The Definitive Technology speakers performed well as long as the deep bass levels were modest, but were no match for the B&Ws in the low bass. Otherwise, their overall balance was quite similar to the B&Ws'.

The SM 450s performed admirably on recordings with high peak content, which profit from high playback levels—big-band material with prominent brass sections and drum rim shots, for example. With the peak-exercising special effects on *Ein Straussfest* (Telarc CD-80098—one of my favorites, even though it dates back to 1985!) I could actually get slightly more volume from the Definitive Technology speakers than from the B&Ws, because the latter's lower sensitivity caused my amplifier to clip before they reached the 450s' maximum level. However, when I got carried away with the volume control on some of the Telarc CD's very loud low-bass passages, I could overload the 450s severely. The 450s' bass response was quite adequate on most of the material I listened to. On shaped tone bursts, bass response was quite acceptable down to 50 Hz, with usable output at 40 Hz—but not at lower frequencies. Teaming the 450s up with a subwoofer improved the sound significantly, putting the 450s on a more equal footing with the much larger 801s.

On well-recorded female vocals, the 450s did exhibit some slight upper-midrange irregularities, but on high-frequency sibilants they did quite well, reproducing them without harshness, strain, or spittiness. After my lab tests revealed high-frequency response aberrations caused by the tweeter resonance mentioned earlier, I listened to the speakers again, but could hear no problems caused by this. (Although my hearing, at this point, is rolled off in the range of this resonance, I sometimes can detect the subharmonics of such resonances.) The 450s were the full equal of the 801s on male speaking voices.

On the stand-up/sit-down pink-noise test, I heard moderate upper-midrange irregularities when I stood up. With the speakers turned upside down, the sound heard from a standing position matched the on-axis sound more closely. I did perform side-by-side A/B mono listening comparisons between an upright and an upside-down speaker. Differences were much less evident with music than with pink noise.

The imaging and soundstaging of the 450s were excellent. Mono center images were quite stable and did not shift when the recording's frequency content changed. The 450s did extremely well on classical a cappella choral music, reproducing the voices and the room's reverberant sound with great precision.

Considering their reasonable price, good looks, and great sound, I highly recommend the Definitive Technology StudioMonitor 450 speakers for stereo use or for a home theater setup. With a competent subwoofer, they provide real competition for many much larger systems. Their high sensitivity and smooth response will be welcome in any music system.

—Don Keele

Powered Monitor Speaker

Genelec HT210

Genelec Inc., 7 Tech Circle, Natick, MA 01760. Voice: (508) 652-0900. Fax: (508) 652-0909. E-mail: genelec.usa@genelec.com. Web: www.genelec.com. Model HT210 2-way active loudspeaker, $2800.00 each in black; other finishes optional. Tested samples on loan from manufacturer.

Introduction

No audio component is perfect, and speakers are the least perfect of all. The imperfections of other components can be too small for anyone to hear, but speakers—all of them—have readily audible deficiencies. The virtue of "active," or powered, speakers is that their electronics can make those defects far less audible: Dedicated electronic equalizers can minimize the speaker's frequency-response errors. Built-in amplifiers can provide the exact power that the speaker (or, better yet, each driver) requires and, if each driver is powered separately, precise active crossovers can be employed instead of cruder, passive ones. What's more, protective circuitry can be custom-tailored to the drivers it safeguards.

Despite their obvious potential for improved performance and reliability, active loudspeakers have never taken off in the home market, probably because most audio consumers already have receivers with a full complement of channel power, or even a stack of amplifiers. Why buy power again?

The complexity of modern audio and home theater systems may change that. In the days of stereo, all you needed was a record player, a tuner, a tape deck, a preamplifier, and a stereo power amplifier, plus five shelves to hold everything. Today, you might have seven source components, a preamplifier, satellite receiver, and an equalizer (well, I do). Who has rack space for an additional seven or eight channels' worth of amplifiers? I sure don't.

So I use active speakers throughout my 7.1-channel reference system; they perform better, conserve space, and (because of their driver-matched power levels and protective circuits) let me leave my system in the hands of a friend without coming home to fried tweeters and the smell of melting voice-coil glue.

Genelec is a fairly new name in the consumer market, but this Finnish company's active speakers are highly regarded and widely used in pro sound, where active speakers have long been common. Now, the company is angling for consumer sales, with several series of active home-theater speakers. The HT210 is the larger two-way speaker system in the Intimate Home Theater series (there's also a three-way system), recommended for rooms of 3,000 to 4,200 cubic feet; other series are designed for rooms of under 3,000, 5,000 to 10,000, and over 10,000 cubic feet. The line also includes an in-wall model and two subwoofers.

The HT210 has a 10-inch woofer, which is unusual in a two-way speaker. The primary reason you don't see many 8-, 10-, or 12-inch two-way designs is that the directivity of large drivers narrows rapidly as they reach the crossover point, which is also the frequency where a tweeter's directivity is widest. Passive crossovers can have little or no influence on directivity, especially while retaining smooth response off axis. Better-performing satellites use 6- or 6½-inch woofers at most, because such drivers are the largest ones capable of offering both a reasonable low-frequency extension and a woofer directivity that closely matches the tweeter's near the crossover (1.8 kHz). With electronic crossovers the designer can play a few little trade-off games regarding directivity; in the case of the HT210, however, the excellent directivity over the entire operating range from 42 Hz to 22 kHz, despite the comparatively large woofer, appears to be due to the shallow, hornlike "Directivity Control Waveguide" surrounding the tweeter.

The HT210 has two internal amplifiers: a woofer amp with a "short-term" power rating of 180 watts, and a tweeter amp with a 120-watt "short-term" output rating. Genelec doesn't say what the hell "short-term" watts are, but who cares? Amplifier power ratings for active speakers (powered subwoofers included) have no signif-

Manufacturer's Specifications

Free-Field Frequency Response: 42 Hz – 22 kHz, ±2.5 dB

Amplifier Power ("short-term"): bass, 180 W; treble, 120 W

Maximum Peak Acoustic Output Per Pair: 124 dB SPL at 1 meter

Dimensions: 19½" (height) x 12⅝" (width) x 11⁷⁄₁₆" (depth) = 495 mm x 321 mm x 290 mm

Weight: 48 lbs. (21.7 kg)

icance. We need to know how much energy comes out of the speaker, not how much energy goes in to produce that output. (Of course, if manufacturer X gets 120 dB SPL with a 2,000-watt amplifier and manufacturer Y does it with 20 watts, I might prefer the latter because it's easier on my electric bill.) What is significant is that Genelec specifies peak output for a pair of HT210s as 124 dB SPL at 1 meter with "music material."

Unlike the controls on passive speakers, the Bass Tilt, Bass Roll-Off, and Treble Tilt controls in the HT210's electronics work almost exactly as specified, even at low frequencies. An "Autostart" function turns the unit off if no signal has been present for 5 minutes, but restarts it immediately when a new signal is received. Additional controls on the rear panel (wouldn't remote controls be cool?) include an on-off switch, a 110/220-volt mains selector, XLR and RCA input jacks, and a rotary control for matching input sensitivity to the output levels of upstream components. Units currently in production have two additional features: a set of contacts for on-off switching using 12-volt trigger signals, and switches that control the LED indicators. (Users will be able to select whether the LEDs remain off, show only yellow for standby and green for operation, or also show red for overload.) The speaker is magnetically shielded, so you can use it near a TV set or other cathode-ray tube (CRT) display.

The HT210 is relatively large for a satellite speaker. (With its bass response specified as -2.5 dB at 42 Hz, the HT210 could conceivably be used without a subwoofer, but I think few would use it that way.) Although it has a small, 1-foot-square footprint, the cabinet occupies 2.8 cubic feet of space in a listening room, and its 48-lb. weight means that you won't be hoisting these speakers off their stands with one hand while dusting with the other. The MDF cabinet of my samples had a flat black pro-style finish, and lacked the optional ($79) grilles. In speakers selling for $5,600 a pair, the utility finish and the extra charge for grilles were disappointing. However, HT210s are now available in glossy piano black and three wood-veneer finishes, all complete with grilles (prices not established at press time). As I did not have the grilles, I could not measure what effect, if any, they'd have on the sound.

The tweeter waveguide plate can be removed and rotated 90° so the Genelec logo will be upright if you mount the speaker horizontally. The electronics panel on the rear is resiliently mounted, a pro-sound carry-over that protects the system against rough handling on tour. The enclosure seems relatively tourproof, too: when I accidentally knocked the HT210 off my measurement stand, the 6-foot drop left only a -by-2½-inch gouge on the rear corner of the cabinet and did not affect the speaker's operation.

Lab Tests

How did the Genelec HT210 measure up? Let's discuss how it performed in the lab first. Basic measurements were taken at 2 meters in my large, 7,600-cubic-foot, room; maximum output for a stereo-arrayed pair was measured at 4 meters in the same room. All measurements were taken with a DRA Laboratories MLSSA acoustic analyzer and an AudioControl SA-3050A third-octave real-time analyzer and sound-level meter.

The horizontal response graph (Fig. 1) shows that the HT210 is incredibly smooth out to 60° off axis. Directly on axis, its response fits in a ±3 dB window from 55 Hz to 20 kHz, shelved up by approximately 2 dB between 1 and 10 kHz. Horizontal directivity is remarkably smooth and wide. This is not due to some kind of electronic trickery; there is no suggestion to that effect in Genelec's specs and literature. As it is most unusual for a 10-inch two-way system to work this well off axis, the explanation probably lies, as I suggested earlier, in the shallow, hornlike baffle ("Directivity Control Waveguide") of the tweeter.

Vertical radiation patterns (Fig. 2a/b) are less uniform. Below the axis, there's a sharp, deep notch at 1.6 kHz, followed by irregularities at greater ra-

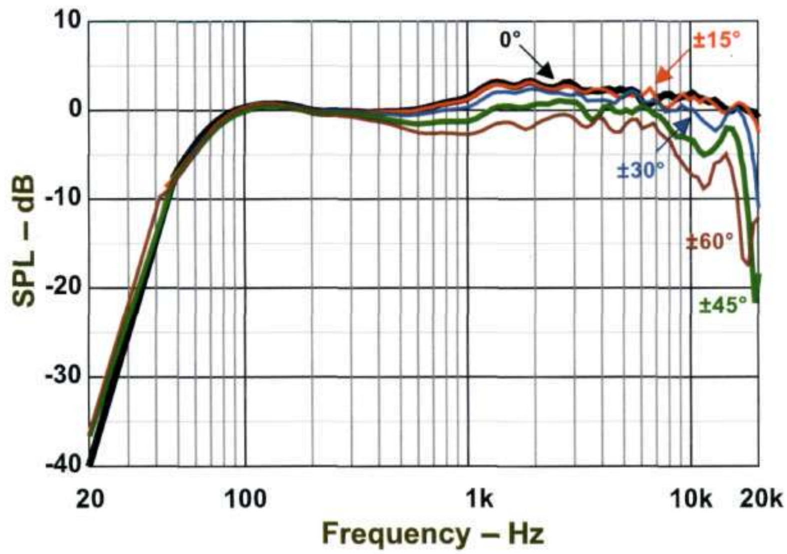


Fig. 1: Horizontal on- and off-axis frequency responses.

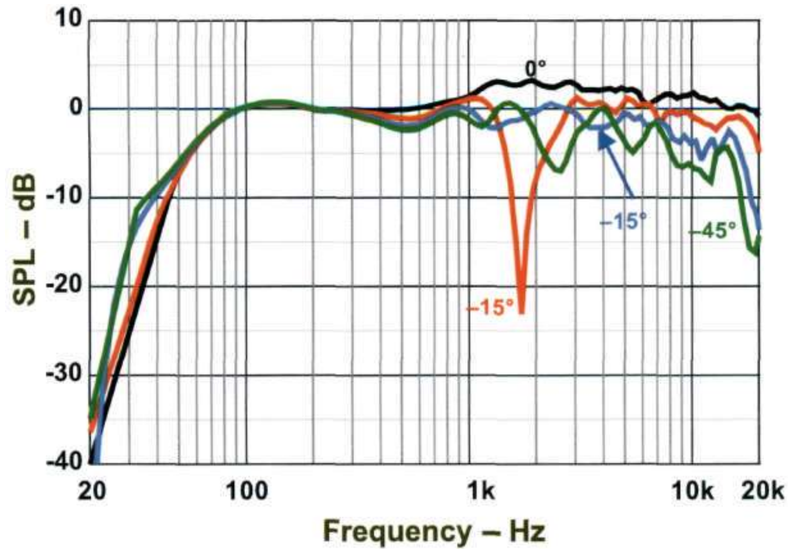


Fig. 2a: Vertical off-axis frequency response below axis.

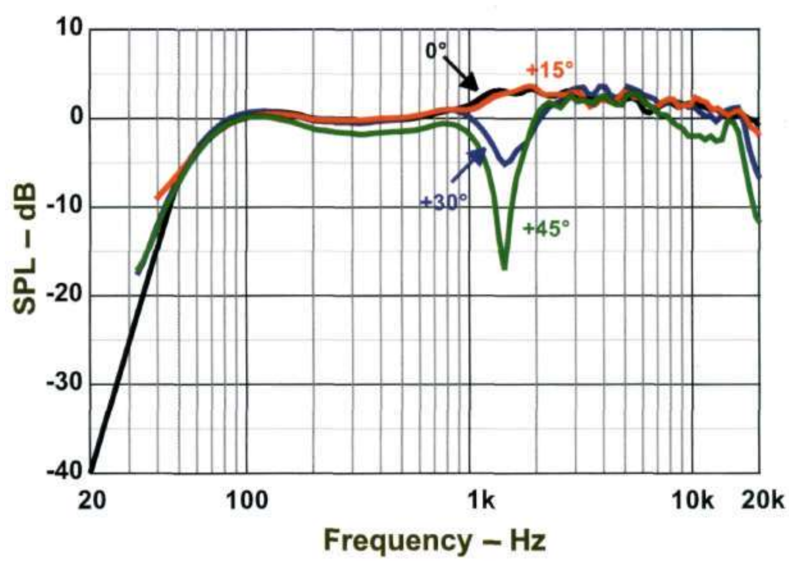


Fig. 2b: Vertical off-axis frequency response above axis.

diating angles. Above-axis response is smooth to about 20°, with notching near the crossover frequency as the angle increases. (These problems are common when multiway speakers have drivers placed side by side or when vertically arrayed systems are used horizontally. I beg people with multiple listening seats to use a vertically arrayed center channel.) The HT210 should be used vertically whenever possible, and when used for a center channel should preferably be placed below the screen.

The response alterations imposed by the Bass Roll-Off and Bass Tilt switches followed almost exactly the curves printed in the manual and on the electronics panel on the back of the enclosure, although the magnitude of action was only about 65% of that indicated. For example, the DIP switch for Bass Roll-Off (a highpass filter whose slope increases from 6 to 12 dB per octave in small steps) indicates cuts of 2, 4, 6, and 8 dB for frequencies below 100 Hz, but setting the switch at -8 dB only cut response by only a little more than 5 dB. Likewise, the Bass Tilt switch (which should cut 2, 4, or 6 dB below 1 kHz, depending upon its setting) produced a 4 dB reduction when set in the -6 dB position.

On the other hand, the action of the Treble Tilt switch, which cuts in at about 8 kHz and was indicated as +2, -2 and -4 dB at 15 kHz, matched the printed graphs exactly. The tweeter and woofer can be turned off individually when the Mute position on the driver's DIP switch is selected—while this is a fantastic feature for nearfield measuring it is of no use I can think of for home listening.

For a two-way satellite, the HT210 delivered a healthy output, though not quite as healthy as suggested by Genelec's specification (124 dB peak per pair at 1 meter, with music). Using the most challenging recordings I have, I got the HT210s to crank out a clean 102 dB SPL peak (no audible amplifier clipping, limiter action or speaker distress) at 4 meters in my large room, which translates to 114 dB at 1 meter. Turning the gain up from the default setting (-6 dBu) to its maximum (+4 dBu) enabled the speakers to deliver 105 dB SPL at 4 meters; however, limiter action and/or amplifier clipping was clearly evident at sound pressure levels above 102 dB; this surprised me, as most active speakers will not allow themselves to be driven into overload.

The HT210's low-frequency abilities were similar to those of many "full-range" floor-standing loudspeakers I've used. Speakers seldom have the low-frequency dynamic capability that reference measurement levels imply. Frequently, full-range models whose measured low-frequency extension seems impressive exhibit an upward spectral balance shift at high output. This shift occurs because the low-frequency driver lacks the displacement to keep up with the mid/tweeters; the highs keep getting louder while the lows stall out as the system's output level increases.

To measure the Genelec's low-frequency abilities, I used a technique adopted from Don Keele: I fed the speaker ramped, 6.5-cycle tone bursts at -octave frequencies, and used a MLSSA acoustical measurement system to determine the maximum low-frequency SPL the speaker could deliver at 2 meters (a truly practical listening distance) before distortion reached 10% or overload protection cut in. While a distortion figure of 10% seems quite high, a speaker still sounds clean at that level. This is because the speaker is just leaving its linear output range at that point; as the level increases further, distortion will begin increasing exponentially. (Technically, this happens when the driver's motor BL product, a measurement of magnetic field strength, has fallen to 70% of its rest-position value, or when the suspension has stiffened by a factor of four.) I define a speaker's bass limit as the lowest frequency and highest SPL it can deliver within the 10% distortion threshold. For a single HT210, the bass limit was 75 dB SPL at 40 Hz at 2 meters.

Surprisingly few two-way satellites (or even full-range speakers) can deliver such usable output at 40 Hz. However, the HT210's usable output at 40 Hz was nearly 25 dB below its maximum clean output at higher frequencies, which occasionally caused the spectral balance shift described previously. If you want full-bandwidth dynamic capability, you'll need to use the Genelec with a subwoofer.

Listening Tests

I listened to the HT210 as a stereo pair. The sound was clean and clear, although somewhat aggressive. With the Treble Tilt switch set to 0 dB, there was excessive sibilance when playing Suzanne Vega's recording of "Tom's Diner" (on *Solitude Standing)* and percussion sounded somewhat overemphasized. When I set the Treble Tilt switch to -4 dB, however, voices and acoustic instruments were rendered with natural timbre and excellent detail and clarity, although the speaker still sounded slightly aggressive.

The Genelecs delivered a wide, moderately deep soundstage, with excellent image placement and separation. The wide, smooth radiation pattern provided an excellent sense of ambience, positioned images outboard of the left/right speaker pair, and clearly rendered reverb and ambient effects in the mix. Center images followed me when I moved off axis; this is normal for two-channel systems, and the Genelecs do a better job of distributing ambience and retaining far left/right images than speakers typically do in stereo setups. I believe the HT210 can be successfully used as a left/right, center, or surround speaker in a multichannel system.

Dynamically, the HT210 plays damn loud, yet retains its clarity when the music gets soft or is simply played softly. There is some, but less than usual, upward spectral shift when playing full-range recordings at very loud levels. When played at full gain with ultraloud, dynamic, or ultra-compressed program material (Radiohead's *Amnesiac,* Fugees' *Blunted on Reality,* Jay Leonhart's *Salamander Pie)* the HT210 could play roughly 3 dB beyond its clean limit. At such high levels, the Genelec's limiters keep turning on and off and the sound is sometimes grossly distorted. (I used hearing protection when checking this.) But when you're finished abusing the speaker, there will be no burned or bottomed voice coils, and the system will play as if it were still new.

Summary

The Genelec HT210 will reward any listener with high-quality, high-output playback in mono, stereo, and multichannel music and film systems. It has more output capability than any other two-way home system I've ever used, and more than many 12-inch towers. As a satellite speaker, it's a little on the large side. As a full-range speaker, it's moderate in size but with the impact of many larger floor-standing systems. Like all satellite and most full-range systems it will benefit from a subwoofer if you like high-impact low-frequency programs.

Some people will consider the Genelec HT210s pricey (a 5-channel system would run you about $14,000) Others, though will see them as a bargain, considering current speaker-price trends and the fact that a full set of high-performance electronics with useful precision operating controls is included in the deal. As far as I'm concerned, these Genelecs would be welcome in my house anytime.

—***Tom Nousaine***

Floor-Standing 2-Way Loudspeaker System

Thiel CS1.6

Thiel Audio, 1026 Nandino Boulevard, Lexington, KY 40511-1207. Voice: (859) 254-9427. Fax: (859) 254-0075. E-mail: mail@thielaudio.com. Web: www.thielaudio.com. Model CS1.6 coherent source loudspeaker, $2390.00 per pair in black ash, cherry, maple, oak, or walnut ($1990.00 the pair painted black). Tested samples on loan from manufacturer.

Introduction

The CS1.6 shares the distinctive look of Kentucky-based Thiel Audio's other floor-standing speakers: a finely finished cabinet with a raked-back, rounded, black front panel. That look is part of Thiel's "Coherent Source" design, which, the company says, aims to eliminate "time and phase distortions that cause alterations in the reproduced musical waveforms of most loudspeakers." Raking the front panel moves the tweeter farther from the listener, so its output will arrive at the same time as the woofer's. The front panel is claimed to reduce parasitic resonances, and its rounded corners minimize diffraction. Thiel also uses wide-bandwidth drivers and true, first-order crossovers to maintain phase coherence. The result, says Thiel, is enhanced realism, clarity, transparency and immediacy, as well as improved imaging and a deeper soundstage.

The CS1.6, the second smallest of Thiel's six CS-series speakers, is a two-way bass reflex system with anodized aluminum diaphragms on both drivers.

The woofer's construction is unusual. Instead of placing a small voice coil at the apex of a deep woofer cone, Thiel gave the CS1.6's woofer a large (3-inch) coil attached about midway between the cone's outer surround and its center. This design distributes the driving force over a larger area and, by reducing the unsupported span between the coil and the cone's edge, reduces cone breakup. According to Thiel, it also moves the diaphragm's spurious resonances to a much higher frequency, and hence raises the driver's high-frequency cutoff.

As a result of this driving system, the woofer's cone is quite shallow and its dustcap is distinctively large. The large voice coil enables Thiel to place the neodymium magnet inside the pole piece rather than outside it. This topology provides magnetic shielding; when I set a CS1.6 right next to my computer's monitor, it caused no color distortion of any kind.

The woofer's extended response is a necessity, because of the CS1.6's first-order crossover. The virtues claimed for first-order crossovers, which have gentle slopes of 6 dB per octave, are simple construction (typically, one capacitor and one inductor) and "phase coherence" (the elimination of phase changes at the crossover frequency). The theory is that a first-order crossover keeps the two drivers in quadrature (90° apart) at all frequencies, and consequently the sum of the two drivers' acoustic outputs is theoretically a perfect replica of the crossover's input. The importance of this from the standpoint of audibility has long been debated and belongs in another discussion.

It's not enough for a first-order speaker system to have crossovers with 6-dB/octave slopes. It's also necessary to have loudspeaker drivers that operate cleanly for two to three octaves beyond the crossover point. This is because moving-coil drivers are second-order devices, which roll off at 12 dB per octave outside their natural passband. Only very wideband drivers allow the system's roll-off to start well before the drivers'. A first-order crossover's gentle slope also does little to suppress any irregularities in the driver's response outside its passband. So using such drivers not only requires an extended upper range for the woofer, but also a downward extension of the tweeter's response. Thiel says quite a bit about technologies that extend the response of the CSl.6's woofer upward, but nothing about the low end of the tweeter's response. Nonetheless, I neither heard nor measured any of the anomalies I'd expect if the tweeter lacked low-end response or had insufficient power handling at low frequencies.

Despite the appealing simplicity of basic first-order filter design, it often

takes sophisticated networks to maintain a true first-order response several octaves above and below the crossover frequency. Thiel's literature says that the company's speakers "make extensive use of network compensation. Typically, about 40% of the network elements are used to achieve correction of what would otherwise be minor response irregularities."

The CS1.6 has an unusual bass-reflex port: a half-inch-wide, 12-inch-long slot exiting through a beveled recess on the front panel. The recess flares outward rapidly to a width of 4½ inches at the panel's front surface. Thiel says this design reduces unwanted port noise, and my experience with the speaker backs that up.

According to the company, the port design also reduces "grille loading effects," and—hallelujah!—the grille had virtually no effect on the sound of the CS1.6. However, that may have more to do with the grille's design than the port's. The grille is a fabric-wrapped sheet of ultra-thin steel, 80% perforated. Magnets hidden beneath the cabinet surface hold the grille to the front baffle, eliminating grille frames and other constructions that could affect response. That baffle, by the way, is 2 inches thick, and the other enclosure panels are an inch thick, stiffening the cabinet and minimizing secondary radiation.

The cabinet is not only stiff but beautifully finished, something Thiel is known for. Buyers have their choice of 15 fine cabinet finishes, at several price levels. In the standard finishes (ash, black ash, cherry, maple, oak, and walnut veneers), the speakers are $2390 per pair. More exotic finishes (such as ebony, dark cherry, mahogany, teak, and zebra wood) are available at prices up to $2865 per pair. At the other end of the scale, a pair of CS1.6s in plain black paint is $1990. Custom finishes are also available. My test sample was finished on three sides and the top in amberwood, an aggressively grained walnut ($2565 per pair); the front baffle and bottom were black.

The CS1.6 comes with four threaded and pointed feet, which can be used to steady the cabinet on a thick carpet and to control the tilt of the front baffle. An optional outrigger base is also available ($200 per pair) for added stability on deep-pile carpets or in homes with active small children or large dogs.

Connections are made via heavy-duty multiway binding posts on the rear panel. They are not on ¾-inch centers, so they won't accept double banana plugs, but single banana plugs work fine. (By the way, you can easily convert a dual banana plug into a pair of singles with diagonal cutters.) There is only one pair of connecting posts per speaker, because Thiel doesn't believe biamping or biwiring are necessary (neither do I). But the company also says that it does not use separate woofer and tweeter connections because the strap needed to bridge die two together for use with just one amp would be "sonically compromised;" that just plays to existing audiophile myths.

Warranty is 10 years to the original owners for any defects in material and workmanship.

Manufacturer's Specifications

Type: 2-way, floor-standing, with ported woofer

Drivers: 6½" aluminum-cone woofer with 3" voice coil, 1" aluminum-dome tweeter

Crossover Frequency: 3 kHz

Rated Bandwidth (–3 dB): 48 Hz to 20 kHz

Rated Frequency Response: 50 Hz to 20 kHz, ±2 dB

Phase Response: Minimum ±10°

Rated Sensitivity: 90 dB at 1 meter, 2.83 Vrms applied

Rated Impedance: 4 ohms, nominal; 3 ohms, minimum

Recommended Amplifier Power: 50 to 300 W

Dimensions: 35½" (height) x 9" (width) x 11½" (depth) = 902 mm x 229 mm x 292 mm

Weight: 38 lbs. (17.3 kg) each

Measurements

My basic measurements of die Thiel CS1.6 were quasi-anechoic, taken at a distance of 2 meters in a 7,600-cubic-foot room, combined with near-field measurements for the low frequencies. All measurements were taken with a DRA Laboratories MLSSA acoustic analyzer with calibrated microphone and an AudioControl SA-3050A third-octave real-time analyzer and sound-level meter. I also supplemented the quasi-anechoic measurements with 2-meter readings taken with the speaker standing on a carpeted floor to replicate normal use. In my opinion, measurements of performance on a floor should always be used when designing and evaluating a tower speaker, because its proximity to the floor will affect its response in any listening room. (Floors also affect the response of stand- or shelf-mounted speakers, but the effects will vary with the height at which the speaker is mounted.)

The characteristic on-axis response of the CS1.6 is basically flat and smooth, fitting inside a ±2 dB window from its 45 Hz low limit to 8 kHz. At higher frequencies, response slopes gently downward at approximately 3 dB per octave. Off-axis response in the horizontal plane (Fig. 1) is very well controlled to ±30°, with only moderate notching near the crossover at wider angles. Vertically, response deteriorates rapidly above axis (Fig. 2a); so it's absolutely necessary to aim the tweeter's axis at the listener's ears by using the speaker's spiked feet to angle its front panel upward. Radiation below the axis (Fig. 2b) is much smoother but, because the speaker is a floorstander, lis-

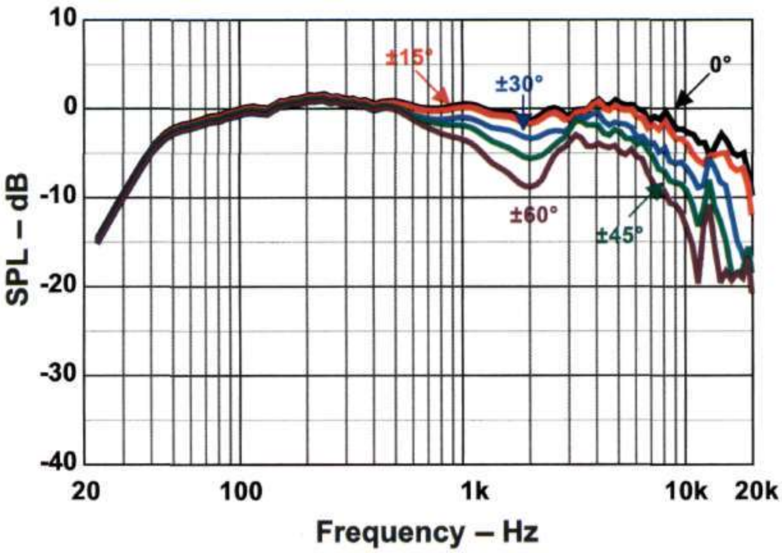


Fig. 1: Two-meter, on-axis (0°) response and horizontal off-axis frequency responses.

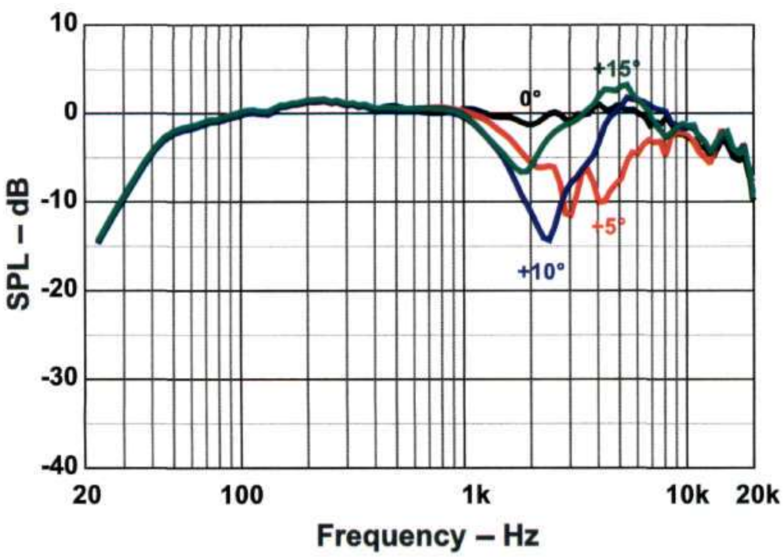


Fig. 2a: Vertical off-axis frequency response above axis.

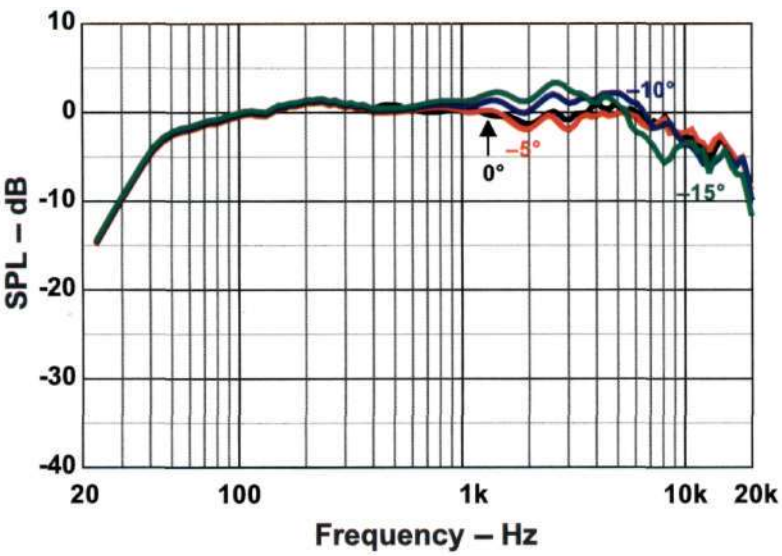


Fig. 2b: Vertical off-axis frequency response below axis.

teners will always experience an octave-wide, 3.5 dB floor-bounce notch centered at 225 Hz. On the other hand, the clever grille has virtually zero effect on the response.

Sensitivity is 90 dB SPL at 1 meter with 2.83 volts applied (equivalent to 1 watt into 8 ohms). The CS1.6 is a low-impedance speaker by any measure. From 200 Hz up, its impedance is 4 ohms or less, reaching its minimum (3.2 ohms) at 5.6 kHz.

I use a ramped 6.5-cycle tone burst at -octave frequencies (adopted from Don Keele) and MLSSA to determine the maximum low-frequency SPL at 2 meters (a useful listening distance) that can be attained without exceeding 10% distortion. While 10% seems quite high, it denotes the point where a loudspeaker is leaving its linear output range (that is, its motor BL product has fallen to 70% of its rest-position value, or the suspension has stiffened by a factor of four). At this point, the speaker still sounds clean, but distortion begins increasing exponentially with further increases in level. I define a speaker's bass limit as the lowest frequency and SPL it can deliver with 10% distortion or less; for a single CS1.6, the bass limit is 83 dB SPL at 40 Hz, measured from a distance of 2 meters.

While the CS1.6 has good extension for a speaker system with a 6.5-inch woofer, aided greatly by the system's 50 Hz tuning, overall bass performance is not especially robust; at 100 Hz, the CS1.6 can produce only 90 dB SPL at 2 meters with less than 10% distortion.

Use and Listening Tests

Two-channel listening may be officially dead in this home-theater era, but I listened to the CS1.6s as a stereo pair so I could better hear what they were doing. Even stereo can mask some operating deficiencies, and Floyd Toole makes a good case for using a single speaker system for listening evaluations.

Spectral balance and dynamics can surely be tested in this way. Certain spatial characteristics, such as openness (the ability to make sound seem independent of its actual source) and reproduction of ambience, can also be fairly evaluated with a single channel. However, a goodly share of spatial rendition is dependent on the room and the positions of the listener and speakers, so I tend to be skeptical of spatial impressions reported in reviews (including mine). So should you.

No matter, my evaluation was conducted with the CS1.6 pair set up well away from side walls and 4 feet from the rear wall, with the speakers 9 feet apart and 10 feet from the listening position. (Thiel recommends a minimum listening distance of 8 feet.) The owner's manual suggests "straight-ahead" speaker aiming. I tried that and "cross-fire" toed-in aiming. For the most part, cross-fire aiming tended to improve perceived spaciousness, tighten image precision, and make the thin bass sound fuller. Of course, much of this is undoubtedly associated with the acoustics of my listening room. Your mileage may vary.

Using the adjustable feet, I tilted the speaker carefully to aim it at my ears. Even though the baffle slopes back a bit more than 10°, the tweeter axis was still somewhat below the plane of my ears at a 10-foot listening distance without this adjustment.

Spectrally, the system had very good clarity and detail but sounded a little thin and soft in the bass; that's often the case with 6.5-inch two-way systems. For example, on Oscar Peterson's "You Look Good To Me," the piano, soft percussive details, and bassist Ray Brown's low-level mutterings were clearly rendered and had natural timbre, but Brown's acoustic bass was somewhat subdued. Likewise on Carla Bley's "Copyright Royalties," the smallest details of brushes, brass, and clarinet sounds were clearly identifiable. Female vocals—by Jennifer Warnes ("Famous Blue Raincoat"), Tracy Chapman ("For You"), and Joan Baez ("Diamonds and Rust")—were also rendered with extremely natural timbre.

Although the CS1.6 delivered center images to centered listeners as well as any two-channel system I've heard, left and right images tended to cluster at the left and right speakers; there was a lack of good mid-left and mid-right imaging. The Thiels seldom managed to convince me the sound was not coming directly from the speakers. When I moved off-axis, the center image followed me; all 2-channel speaker systems are somewhat subject to this effect, but not always to this extent.

On recordings with loud bass, the CS1.6's spectral balance shifted upward as volume increased. For example, the strong bass line on Roy Orbison's "Dream You" just disappeared at high levels, and the percussion on *The Sheffield Track Record* and Joe Farrell's kick drum in "Upon This Rock" went 'pop, pop' instead of delivering a solid whack to the sternum. Similarly, when turned up to high levels, Heart's "Magic Man" tended to shriek.

The pair of Thiels did, however, sound clean, with no obvious distortion and only modest spectral shift at levels up to 99 dB SPL at a 10-foot listening distance in a large room. On material with very low frequency content (Bass Connection's "Drivin' Bass"), overload would occur at 80 dB SPL, but with remarkably little noise from the port. These dynamic limitations are of limited importance for jazz, soft rock, middle-of-the-road music, and all but the most challenging classical material—but I wouldn't recommend the CS1.6s for hip-hop or heavy metal.

Summary

The Thiel CS1.6 is best suited for listeners who listen to traditional two-channel jazz and classical music in a moderate-sized room. Adding a subwoofer will generally make the system useful with a wider range of more dynamic programming, but it will never be suitable for head-banging.

Home theater? By themselves, the Thiels lack the loudness and bass for that, unless you add a subwoofer—and as long as you're doing that, you might want smaller speakers anyway. And somehow, I can't see their styling fitting gracefully into a multichannel system.

What should we expect from a two-way speaker in the CS1.6's price range? Excellent cosmetics? Thiel nails it; great finish, state of the art grille. Excellent spectral presentation? Right on, Thiel. A design philosophy and interesting tech story that will keep owners loving these speakers through the years? That's here in spades. State-of-the-art dynamics? No—not with a 6.5-inch speaker, no matter how well-respected its maker.

In all, the CS1.6 is an excellent-sounding speaker with limited dynamics. That's about all you can expect from any small-woofer system. Yes, there are systems that work nearly as well for a lot less money. But, I can't think of any that have nearly this nice a finish.

—***Tom Nousaine***

Floor-Standing 2-Way Loudspeaker System

Ohm Acoustics Walsh 200 Mk-2

Ohm Acoustics Corp., 76 Degraw Street, Brooklyn, NY 11231. Voice: (800) 783-1553 or (718) 422-1111. Fax: (718) 422-0076. E-mail: OhmSpeakers@aol.com. Web: www.ohmspeakers.com. Walsh 200 Mk-2 floor-standing 2-way tower speakers, $2595.00 to $2995.00 per pair, depending on finish. Tested samples on loan from manufacturer.

Editor's Note: *The origin of the unique Walsh driver appears to have been forgotten, or at least allowed to lapse into some kind of vague folklore, over years; the following is the "official" version, which I can confidently tell—because I was there.*

Ohm Acoustics was founded in 1971 by Martin Gersten, a self-taught loudspeaker designer, for no other reason than to become employed again after he had lost his job at Rectilinear (a speaker company that went out of business many years ago). Gersten had several silent partners with a financial interest in the new company; I was one of them. (I was still in the advertising business at the time; three years later I sold all of my stock back to the company, so for the past 29½ years I've had nothing to do with Ohm—-just to reassure the three or four aging conspiracy theorists who still haven't given up trying to catch me in a conflict of interest.) At first the company made only conventional "monkey coffins" (as rectangular box speakers were condescendingly called in those days), but soon Lincoln Walsh, whom Gersten had known for some time, entered into negotiation with Ohm to have his patented loudspeaker invention developed, manufactured, and marketed. (I would never have become involved with Ohm—whose name was actually my suggestion—if I hadn't known that the Walsh speaker was coming)

Lincoln Walsh was a veteran engineer, a member of the team that had developed radar during World War II and the designer of the legendary Brook triode amplifier (circa 1947). His loudspeaker invention was based on a simple insight: No speaker cone is actually a piston, in the sense that its perimeter moves the same instant as its apex is set in motion by the voice coil. It takes a finite amount of time for an impulse to travel from the apex to the perimeter. A good woofer cone appears to be a piston only because the wavelengths it reproduces are so large that the transmission time from apex to perimeter represents only a tiny fraction of the wavelength and does not result in a perceptible ripple or breakup. A cone reproducing the full audio range, however, inevitably ripples and breaks up, because the higher-frequency wavelengths are only inches and the cone is relatively large, requiring several cycle durations for the signal to travel from apex to perimeter. This is true of all cones, regardless of cone material or geometry. They are, in effect, transmission lines, albeit poor ones.

So Lincoln Walsh said, "If you can't lick'em, join'em!" If a speaker cone is not a piston but a transmission line, let's make it a good, well-organized transmission line! He inverted the driver and turned it apex up, so it fired downward into the enclosure, with the sound coming off the convex side of the cone. He made the cone material stiff, so that sound waves were transmitted in it at a calculated speed that was much higher than in air, and he made the slope of the cone exactly such that the horizontal vector of the transmission synthesized a coherent cylindrical wave front in the air, starting at the cone surround (Fig. 1). One cone covered the full audio spectrum, omnidirectionally, without crossovers and without any interference with the original waveforms as seen by the voice coil. (Note that any old inverted, downward-firing cone driver is, when you think about it, a Walsh driver, just a very bad one.)

This, of course, is a simplistic summary of the theory behind invention; in the real world there were huge problems—efficiency, cone material issues, resonances (and what resonances!), inadequate termination of the transmission line at the surround, etc., etc.—as Ohm soon found out. The Walsh design then went through numerous experimental and production models. In my opinion, the perfect Walsh driver has yet to be made; if it were made, it would be the world's simplest, most beautiful, most unproblematic speaker design. (Perhaps German Physiks, a

Manufacturer's Specifications

Type: 2-way, floor-standing loudspeaker

Drivers: One 10″ deep-cone full-range driver and one 1″ dome tweeter

Crossover Frequency: 8 kHz, no other details given

Rated Frequency Response: 25 Hz to 20 kHz, ±3 dB

Rated Sensitivity: 87 dB at 1 meter, 2.83 Vrms applied

Rated Impedance: 6Ω nominal, 4Ω minimum

Recommended Amplifier Power: 80 to 220 W (maximum dynamic peak power: 400 W)

Dimensions: 42¼″ (height) x 11¾″ (width) x 15¾″ (depth) = 1073 mm x 298 mm x 400 mm

Weight: 74 lbs. (33.6 kg) each

Available in real wood veneers

Frankfurt company, has come closest to it, at a very steep price, since the expiration of the patent—but that's another story.) Ohm Acoustics, whose ownership shifted several times over the decades and is now headed by John Strohbeen (formerly of the defunct Tech HiFi chain), ended up making the speaker neither full-range nor omnidirectional—that's how they got around the design challenges. Lincoln Walsh may be a little restless about that in his grave (he died in the early 70s), but at least the major problems have been eliminated.

—Peter Aczel

Acoustic barriers now partially surround the downward-facing driver to reduce rear sound radiation, and a tweeter has been added. Both changes tailor the speaker's lateral coverage so that listeners toward the left of the room will hear more of the right speaker than they would from conventional speakers, and listeners toward the right will hear more of the left speaker. This compensates for the precedence effect, which makes the stereo image collapse toward the nearer speaker (whose sound arrives first and is louder) for listeners who are not equidistant from both. The tweeter (which I consider a supertweeter, as it operates only from 8 kHz up) also augments the high-frequency response. Ohm does, however, offer a few models in the original omnidirectional format, for special applications such as surround channels or background music.

Ohm offers Walsh speakers at prices from $ 1395 to $4495 per pair (plus shipping). The company suggests using the 200 Mk-2's only for rooms up to 15 by 25 feet (375 square feet). For rooms up to 25 by 32 feet (800 square feet), the 300 Mk-2's are recommended, while the smaller 100 Mk-2's are for rooms 14 x 20 feet (280 square feet) or smaller. According to Ohm, all these speakers have such robust bass that subwoofers are not needed. All Ohm speakers are sold factory direct, with a generous two-month home trial program.

The Walsh 200 Mk-2 is a direct descendant of the Ohm Walsh 4 sold in the 1980s. Our review sample, in fact, started out as a Walsh 4. This was possible because Ohm provides an upgrade program for older systems; Ivan Berger's sidebar tells what it's like to upgrade an older model.

Ivan's upgrade involved replacing the drivers and crossovers, which are built into a squat, perforated-metal cylinder roughly 9¼ inches in diameter and 8 inches high. The two drivers within the cylinder are a Walsh driver, 10 inches in diameter, and a dome tweeter. The head assembly was so well put together that I was not able to take it apart for observation without damaging it irreversibly, but there is an illustration on Ohm's Web site (http://www.ohmspeakers.com/coherentlinesourcedriver.cfm) that shows what's inside. The inverted conical surface of the cone driver radiates all the system's sound omnidirectionally except in the top octave, where the small dome tweeter mounted on top of the Walsh driver's magnet assembly takes over. When the speakers are set up in a normal listening configuration, the tweeter is aimed 45° laterally off the frontal axis of the system towards the inside space between the speakers. An oversized, trapezoidal space frame made of metal and covered in grille cloth fits over the cylinder and completely covers the top of the system.

The cabinet of the 200 Mk-2 is a straight-sided, vented enclosure that's deeper than it is wide. It's constructed of 16-layer birch plywood covered with real wood veneer on all four sides. Ample internal cross-bracing increases the cabinet's rigidity. The vent is a port tube, 2¾ inches in diameter and 15 inches long, or about half the cabinet's height; the tube exits through a hole on the cabinet's bottom. Signal connections are made via a single set of gold-plated, double-banana five-way binding posts, also on the bottom. The posts can handle wire up to a generous 0.22-inch diameter (AWG #4).

Four furniture casters make it easy to move the Walsh 200 Mk-2 around. Why roll-around casters rather than spikes or feet? Ohm believes you'll get the best from speakers if you can easily experiment to find what location and orientation optimize imaging and bass response. Ohm points out (and I emphatically agree) that the very audible changes that result from repositioning speakers make a far bigger difference than the subtle changes that occur when using different types of cabinet feet. Also, there is no possibility of the cabinet's moving back and forth during

Fig. 1: In a Walsh driver, waves traveling down the cone produce a cylindrical wave front.

loud passages, because the woofer moves only up and down.

Measurements

I measured the performance of Ohm Acoustics' Walsh 200 Mk-2 in my usual way: making nearfield and ground-plane measurements to assess die speaker's low-frequency response and using windowed, in-room tests to measure response at middle and high frequencies.

In addition to my customary tests, I did a complete set of horizontal off-axis response curves every 10° completely around the Ohm to investigate its full-circle lateral soundfield. The test microphone was located at a distance of one meter from, and aimed at a point three inches below, the top of the driver's cylindrical cage; a 2.83 Vrms signal was applied to the speaker. One-tenth-octave smoothing was used in all the following curves.

Figure 2a shows various horizontal off-axis frequency responses of the Ohm Walsh. Normally, my first graph is of on-axis frequency response but, with this speaker, where would "on axis" be? Directly in front of the cabinet? At some unspecified horizontal angle to the cabinet's front panel? Someplace else? Only after waving around a microphone connected to a real-time third-octave spectrum analyzer did I get a general idea.

The maximum radiation appeared to be directed laterally at 45° off the box axis towards the center line (inside) between the speakers, i.e., the left stereo speaker directed its sound 45° to the right and the right speaker's sound was directed 45° left. Maximum radiation vertically appeared to be roughly aligned with a point about one-third the way down from the top of the cylindrical cage. The lateral direction and height coincide with the supposed radiating direction and vertical location of the system's tweeter. (There was no way to get into the driver cage and check my hunch. Drat!)

The Ohm's frequency response at various horizontal angles is shown two ways: as conventional response curves (Fig. 2a) and as an "overhead view" of the response contours (Fig. 2b). Consider "inside" as the direction from either speaker of a stereo pair toward a centered listener, and "outside" as the direction from the speaker to the room's nearer side wall.

Figure 2a shows response from 90° outside to 90° inside in 30° steps, plus a 45° inside curve that corresponds to the direction of strongest response. The symmetry of the response curves around this 45° curve is clearly evident. The 30°, 45°, and 60° inside curves essentially lie one on top of the other, which indicates excellent directional uniformity. Note that the 0° and 90° inside curves are also very close to each other but exhibit lower high-frequency output than the other three inside curves. Outboard of the 0° (straight-ahead) curve, there is substantial high-frequency rolloff above 1 kHz, increasing with the angle. (This, and the fact that the main response axis is toed in at 45°, help make early reflections from the room walls less troublesome.) But between 0° and 90° inside, coverage is quite uniform. As far as smoothness and spectral balance are concerned, the 30°, 45°, and 60° inside curves are quite well behaved, except for a slight uptilt in high-frequency response above 14 kHz and a moderate depression in the midrange response between about 400 Hz and 3 kHz.

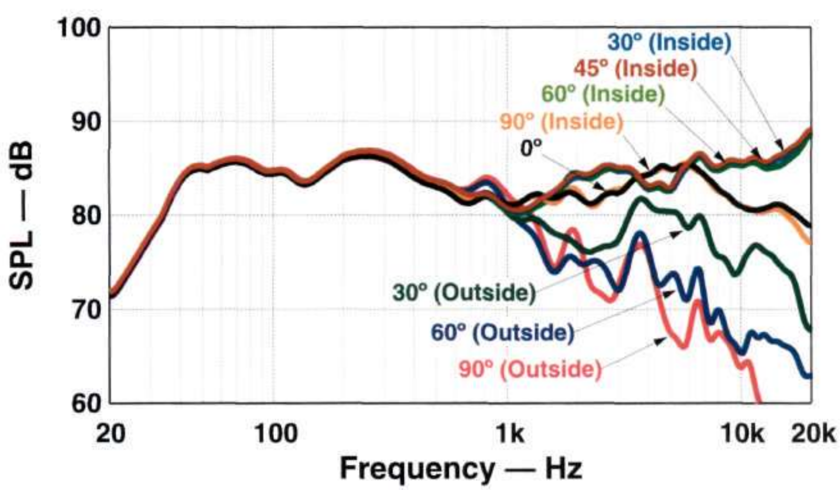


Fig. 2a: Horizontal off-axis frequency response.

Another anomaly is a slight upper-bass/lower-mid response hump centered at 250 Hz, which appeared regardless of the angle or distance at which I made my measurements. However, such direction-independent response anomalies are easy to handle with appropriate equalization. The Ohm's anechoic bass response is well behaved, extending strongly to 40 Hz, then rolling off at 12 dB/octave below that. In a room, response below 40 Hz would be stronger, due to reinforcement by the room's boundaries.

The speaker's sensitivity, averaged from 250 Hz to 4 kHz, was 84 dB; that's 3 dB less than specified by Ohm, primarily because the depression in the system's output between 400 Hz and 3 kHz happened to roughly coincide with my sensitivity measurement span. Right-left matching was good; the right and left speakers agreeing within ±1.5 dB, with most of that difference occurring above 10 kHz, that is, in the tweeter's range. The grille of the Mk-2 had minimal effect on response.

Figure 2b shows the 360° directional response of the Ohm Walsh over the full audible frequency range. (I chose this method of display rather than the usual horizontal off-axis "waterfall" display because it makes the Ohm's radiation pattern more understandable.) The vertical axis displays the horizontal angle from 180° (behind the speaker), around to 0° (straight ahead), and continuing around back to 180° again. Positive angles are to the inside, negative angles to the outside. The horizontal axis displays frequency, from 20 Hz to 20 kHz. The display has been normalized to the 45° inside frequency response, the direction of the strongest radiation; this is equivalent to making measurements after the speaker's frequency response has been equalized to be flat at an inside angle of 45°.

The numbered curves are constant-level contours for levels from -30 to +1 dB. (The top left " - 1 " curve shows, for example, that the Ohm's response is down 1 dB at an inside angle of 135° for frequencies from about 50 to 200 Hz.) Levels at intermediate points can be read from the color coding, which is explained by the color scale at the bottom left of the graph.

Now that we have this fancy colored graph, what does it all mean? If you take slices of the graph along the frequency (horizontal) axis at a particular angle relative to the speaker's axis, you'll see the speaker's frequency response at that angle. If you take slices of the display along the angle (vertical) axis at a particular frequency, you'll see the Ohm's lateral polar response at that particular angle.

If a speaker were omnidirectional in the horizontal plane at all frequencies, the entire graph would be yellow. If the speaker radiated sound only to 45° on either side of its axis, but had a perfect directional radiation pattern between those two angles, the graph would show a horizontal yellow bar that rapidly changed to red and then black at angles beyond ±45°. This would mean that, within ±45°, the speaker had flat frequency response at every horizontal angle, or precisely even horizontal coverage at each frequency. Of course, real-world loudspeakers are not this well-behaved.

Getting back to the Ohm Walsh 200 Mk-2, Fig. 2b makes it even easier

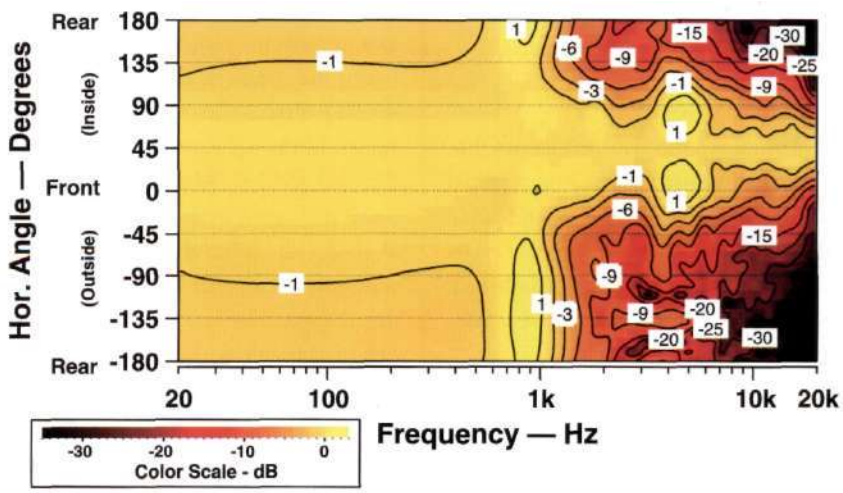


Fig. 2b: Horizontal off-axis frequency response contour plot (response normalized to the response at 45° inside); see text.

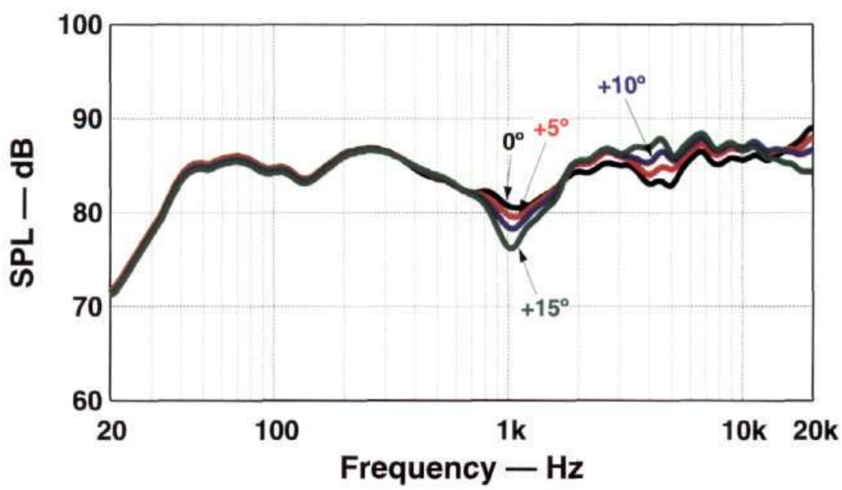


Fig. 3a: Above-axis response.

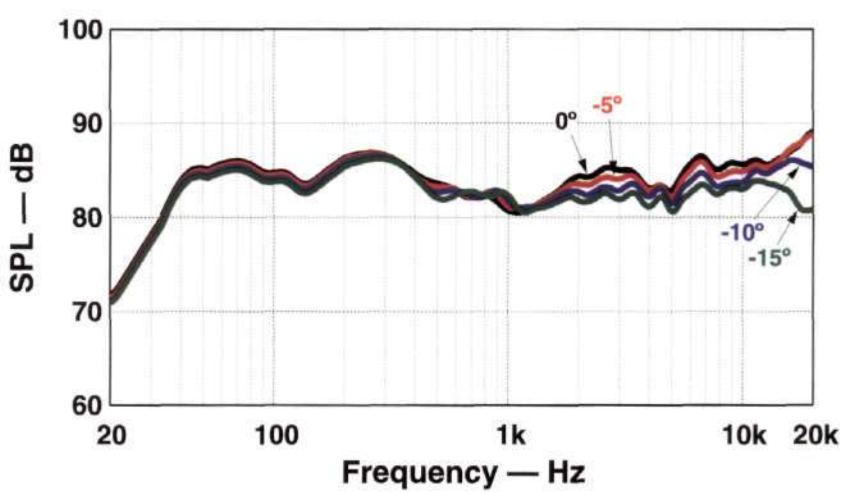


Fig. 3b: Below-axis response.

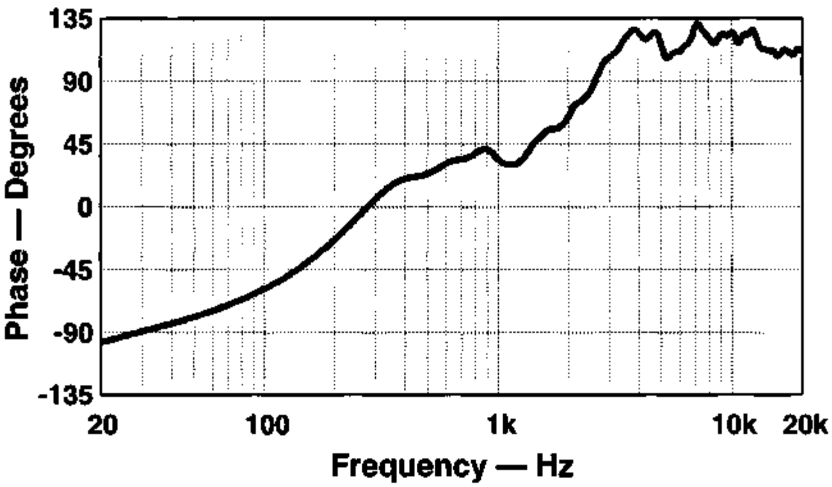


Fig. 4: Phase response at 45° off axis (inside), horizontal.

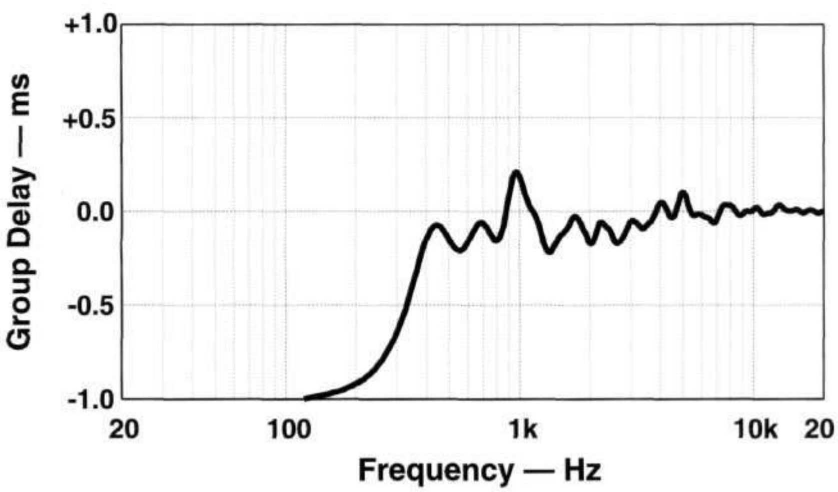


Fig. 5: Group delay at 45° off axis (inside), horizontal.

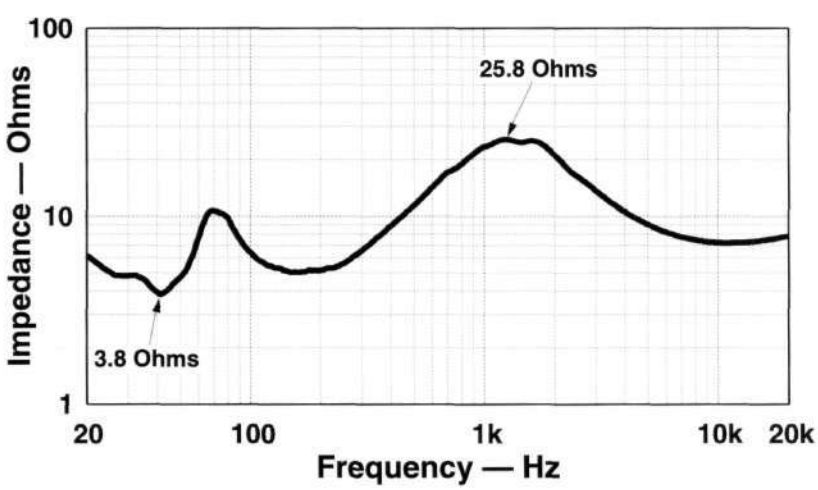


Fig. 6a: Impedance magnitude.

to see that the lateral response is symmetrical about the 45° inside angle rather than about the speaker's axis (0°). In the range from 0° to 90° inside, the response is very uniform over the whole frequency range (the graph is mostly yellow in this range). The speaker is clearly omnidirectional below 1 kHz, but above that frequency its output is pretty much restricted to the range from straight ahead (0°) to 90° inside.

Shown in Figs. 3a and 3b are the Walsh Mk-2's vertical off-axis frequency responses, measured at the 45° inside horizontal angle, from +15° to—15° in 5° increments. Above-axis response (Fig. 3a) is fairly well behaved except for a dip at 1 kHz that deepens at higher angles, and slight irregularities at higher frequencies. Response below axis (Fig. 3b) is even better behaved, with no sign of the 1-kHz dip; at -15° (down) the response is exceedingly flat from 500 Hz up.

To check Ohm's claim that the Walsh 200 Mk-2 preserves waveforms, I measured the speaker's phase and group delay responses at the 45° (inside) horizontal angle. For both measurements, I set the receive delay of my analyzer to coincide with the arrival of the tweeter's signal, which flattens the phase response above 5 kHz, primarily in the tweeter's frequency range.

The measured phase response (Fig. 4) shows that the Ohm is not a linear-phase or minimum-phase system. (If it were, its phase response would be flat and near 0° over the whole frequency range, assuming its amplitude response was also reasonably flat.) However, the Ohm's phase rotates only by about 200° from 20 Hz to 20 kHz, much less than in typical two- or three-way speakers.

The group delay (Fig. 5) also indicates minimal time error. Above 400 Hz, the average group delay varies only by about 200 us (0.2 ms). The irregularities in the group delay are directly due to bumpiness in the amplitude response (the "45°" curve in Fig. 2a). Although the phase and group delay responses do not indicate that the Walsh 200 Mk-2 will preserve waveforms exactly, it will do significantly better in this regard than most competing speakers. Interestingly, the Ohm Walsh driver's output at middle and low frequencies leads the tweeter's output, rather than lagging behind as it would in most two- or three-way direct-radiator speakers. Stated another way, the output of the inverted-cone woofer/midrange reaches the listener about 200 us before the tweeter's output. Spatially, this is an offset of about 2.7 inches and roughly corresponds to the distance between the centrally located tweeter and the outside edge of the woofer/midrange.

Figures 6a and 6b show the 200 Mk-2's input impedance magnitude and phase from 20 Hz to 20 kHz. The impedance magnitude drops to a low of 3.8 ohms at 40 Hz, near the tuning of the vented bass box, and reaches a high of about 26 ohms in the midrange, at 1.3 kHz. The phase is well behaved and covers a moderate range, from about +50° at 500 Hz down to a minimum of about -35° at 3 kHz. The Ohm Walsh should be an easy load for any power amplifier or home-theater receiver.

Below 20 Hz (not shown in either graph) the 200 Mk-2's impedance magnitude continually rises as frequency is lowered and the phase approaches a constant angle of-90°. This indicates that a capacitor is in series with the input of the system, which I verified with an ohmmeter. Why? Because a capacitor very nicely limits input below 20 Hz, where the woofer can be easily overloaded. An excellent addition, especially considering that input capacitors on loudspeaker systems are extremely rare due to the high cost of the high-value, high-quality components required.

The last two graphs (Figs. 7a and 7b) show the harmonic and intermodulation distortion of the Mk-2. The details of the test methods are outlined in my review of Definitive Technology's Stu-

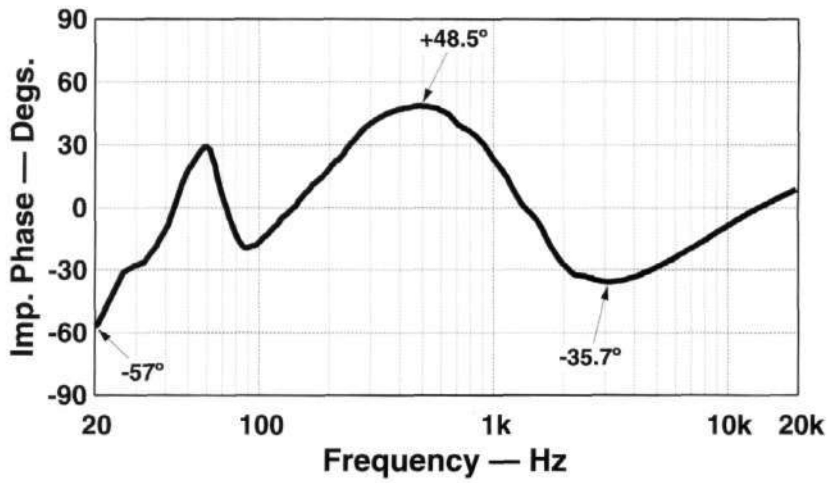


Fig. 6b: Impedance phase.

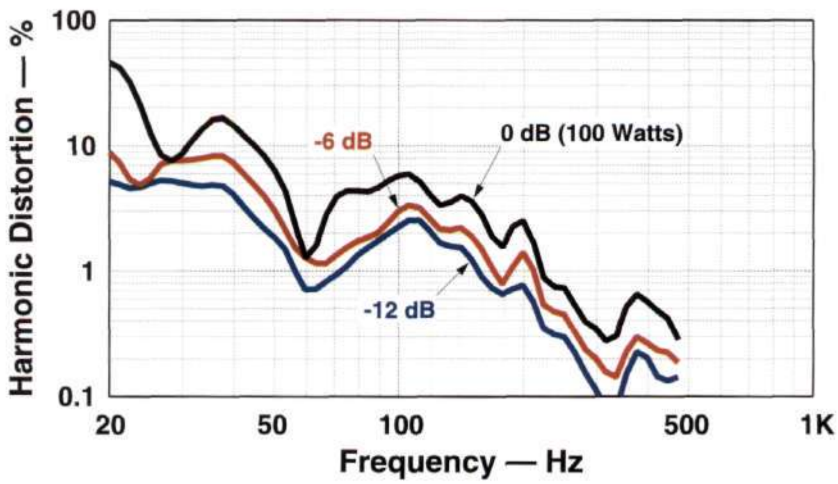


Fig. 7a: Harmonic distortion versus frequency and power level.

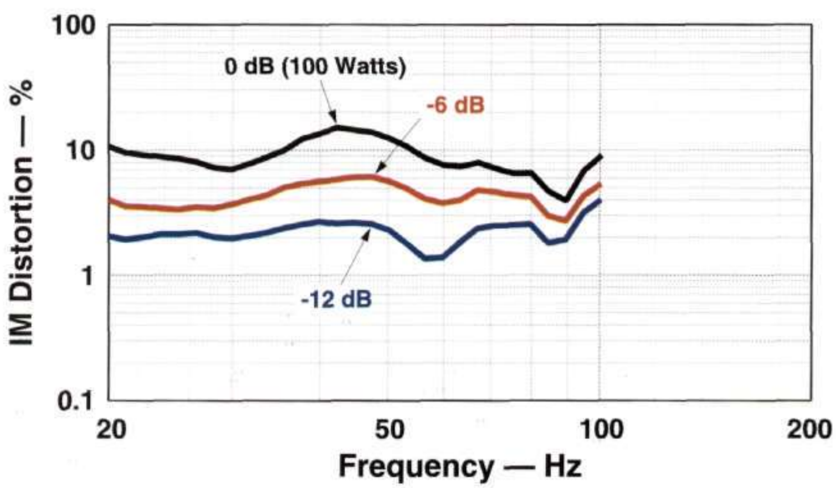


Fig. 7b: Intermodulation distortion versus frequency and power level.

dioMonitor 450, elsewhere in this issue. The distortion of the Mk-2 was measured in the nearfield, with both systems on the floor in the middle of my listening room. The systems were lying on their side, at right angles to each other, with the bottom of one system facing the top of the other, and were driven in parallel. This configuration allowed the acoustic output of the head unit of one system to properly combine with the port output of the other to form an effective overall response measurement.

The sine-wave harmonic distortion of the Mk-2 between 40 and 500 Hz (Fig. 7a) was measured in 6 dB steps at three power levels: 100 watts (0 dB), which corresponds to 20 Vrms into the rated 4-ohm minimum load; 25 watts (-6 dB), and 6.25 watts (-12 dB). The second through fifth harmonics were included in the distortion calculations. At 100 watts, distortion was low, rising only to 6% at 100 Hz, dipping to a very low 1.3% at 60 Hz, and then rising to a still moderate 17% at 37 Hz. The measured distortion rose into objectionable ranges only below 25 Hz. Above that frequency, the 200 Mk-2 sounded quite clean, even at the maximum tested power.

Interestingly, all three harmonic distortion curves reached minimums at about 60 Hz. This behavior would normally be associated with a speaker's vented-box resonance, but the Ohm's resonant frequency is considerably lower. I determined this by measuring the port's output in the nearfield and finding that it reached a peak at 45 Hz, quite close to the 40 Hz impedance dip seen in Fig. 6a. I believe the box tuning is closer to this lower value.

This was just one sign that the Ohm Walsh 200 Mk-2, despite its ported cabinet, does not act like a conventional vented-box speaker. The impedance magnitude curve does not show the usual second peak below the box resonance. More important, though the Ohm's distortion is rising below resonance, it does not rise as dramatically as it would in a traditional vented-box speaker. I believe this is due to the effect of an internal damping blanket stretched across the bottom of the driver cage, just below the woofer-midrange. The air moved by the woofer must pass through this acoustic resistance to reach the inside of the enclosure. This changes the system from a pure vented box into a lossy design, which is somewhat closer to a closed box that doesn't exhibit the rapid rise in distortion below box resonance. The Ohm enclosure's design effectively combines the advantages of a vented box, with its distortion-reducing capabilities at and near box resonance, with the power-handling capability of a closed box at low frequencies.

Figure 7b shows the system's two-tone intermodulation distortion (IM), evaluated at the same power levels as in the previous test. Two equal-level tones, one at 440 Hz and the other swept from 20 to 100 Hz, were applied to the system. The intermodulation sidebands around the higher frequency were evaluated out to the third order, and the test results expressed as a percentage of the energy of the two original test tones. At the highest (100-watt) level, the IM generally stays below 10% over the whole measured range. At lower levels, the IM is correspondingly lower, staying below 5% at the -6 dB level and below 3% at the -12 dB power level. The Ohm's IM was not too objectionable subjectively, even at the 100-watt power level and at 20 Hz.

All in all, the Mk-2 did quite well in the harmonic and intermodulation tests. It performed well all the way down to 20 or 25 Hz.

Use and Listening Tests

The Ohm Walsh 200 Mk-2 speakers were shipped in five separate boxes: one for each cabinet, one for each driver assembly, and one containing the two grilles. This allowed me to unpack and assemble the heavy parts of each speaker separately, which made unpacking and setup a breeze. Assembly consisted of attaching the cylindrical head units to the top of each cabinet with several long woodscrews. Ohm eases this process by providing a screwdriver with extra long shank to clear the top of the head unit, a nice touch.

The head unit is connected to the cabinet by a heavy-duty industrial connector wired with heavy-gauge, audiophile-grade stranded cable. As stated before, connection to the system is via a pair of double-banana jacks mounted somewhat inconveniently on the bottom of the system. I would have preferred having the connectors mounted on the rear, but that would have interfered with the clean look of the cabinet when viewed from behind. Remember that Ohm makes an omnidirectional version of this system (for use in surround channels), which might be visible from all four sides.

Ohm recommends that, for best imaging and smoothest bass, the Walsh 200 Mk-2 speakers should be spaced wide apart, relatively close to the rear wall, and at least two feet from the corner and side walls. Ohm also suggests a laterally asymmetric room placement to further smooth the bass response. For proper imaging, the speakers should be set up so that the front of each cabinet (the side with the Ohm logo) faces straight into the room; this ensures that the speakers' main radiation axes cross in front of the listener. Following Ohm's instructions, I set the speakers up about nine feet apart, spaced them about one to two feet from the rear wall, and made sure that each speaker was facing forward and in the correct channel; the right and left systems are marked with arrows (normally hidden by the grille) that should point towards the center of the room.

Once set up, the Ohms looked quite handsome—tall and slender—even though their appearance is very atypical for speakers, because the grille extends from the top of each cabinet rather than covering about two thirds of its front. The roll-around casters not only added to the

Upgrading Old Ohms

Although some (mostly high-end) electronics can be upgraded to incorporate recent improvements, few speakers can. It's too awkward to ship any but the smallest speakers back to the factory for upgrades, and do-it-yourselfers encounter the problem of removing and replacing glued-in drivers or delving into scratchy fiberglass to remove and replace crossovers.

Ohm's Walsh speakers are an exception. An Ohm Walsh speaker's drivers and crossover are part of a one-piece assembly (the sealed cage that frustrated Don Keele) that's screwed-not glued-in place; this makes installation easy. That leaves a few remaining details, but they're usually simple ones.

My own Walsh 4's being 20-odd years old, I thought it time to bring them up to date; that would be a lot cheaper than replacing my old Ohms with new ones. The update made them almost clones of the 200 Mk-2's in the accompanying review.

At first, the job looked simple. For my Walshes, the driver can is attached to a squarish board that's secured to the cabinet by four extremely large, easy-to-tighten thumbscrews, and a single plug connects the driver can to the cabinet's wiring. To retune the cabinet for the new drivers, I had to slip the old port tube out of the cabinet bottom and replace it with the new one Ohm provided. (The company says this retuning yields deeper bass.) Owners of Ohms less than about 10 years old have the option of prying out the panel that holds the spring-clip input terminals (which accept double-banana plugs) and hot-gluing a new panel, with multi-way binding posts, in its place.

For my ancient Ohm, replacing the panel was not optional. When I took off the old driver assemblies and unpacked the new ones, I discovered that the new assemblies had a two-pin plug and my cabinet had a three-pin connector. That was because Ohm used to mount the crossover on the input panel, but now puts it in the driver cage. It took me and Ohm a while to get that straightened out and for me to receive and install the plugs I needed; luckily, I had other speakers I could use while I waited.

In changing the input panel, I lost one feature of the old Walsh: switches that raised and lowered bass and treble, plus a third switch, "Perspective (Rear/Middle/Front)," that tweaked the midrange level slightly to make the music sound a bit closer or farther away. I regret their loss a little, but I can't say whether I really need them or just miss having something to tweak. Ohm's most expensive speaker, the Walsh 5 Mk-2, retains these controls but Ohm's other models don't. According to John Strohbeen, Ohm's president, these other speakers are designed for specific room sizes, and the Walsh 5 retains these controls "to allow our biggest system to be used in small rooms."

With the drivers hidden within a cloth-lined cage, it's impossible to see when they're facing front. But the thumbscrews and screw holes that attach the driver platform to the cabinet are arranged asymmetrically, ensuring that it can only be mounted with the drivers facing front. At first, the new assembly wouldn't drop into position. Then I realized that my cabinets had warped a small fraction of an inch over the years; when I spread their walls slightly with a screwdriver, the driver platforms dropped into place.

Because so much time elapsed while the mismatched-plug problem was sorted out, I can't really say how well my upgraded Ohms compare to my original pair. But I can say that they perform almost identically to the 200 Mk-2's, which use the same drivers and crossovers. The old and new cabinets have the same enclosure volume, but different construction. My enclosures are of veneered fiberboard, with sloping sides and heavy internal bracing. Current enclosures are of veneered birch plywood, with straight sides and slightly less bracing. That's enough to add a decibel or two of output between 40 and 60 Hz, which would probably be unnoticeable in normal listening.

The price of an upgrade kit varies with the model. Ohm charges $995 per pair for the Walsh 4 upgrade, as long as the old drivers, crossovers, and vent tubes are returned to the company (so factor in some shipping costs). Upgrade prices for other models can be found on Ohm's Web site. For speaker upgrades, Ohm's free home trial period is 60 days long, and its limited warranty on parts and labor runs three years-not as long as for new Ohms, but still generous. Frankly, I'm not much concerned about the warranty, since my original Ohms worked without a hitch for more than 20 years.

-Ivan Berger

tall and thin effect but were a boon when I was moving the Ohms around to optimize their sound. The cabinet's fit and finish were very good, with top-rate coloring and grain on the wood side panels.

I drove the systems with my Crown Macro Reference power amplifier and Krell KRC preamp. I was completely surprised by how different the Ohms sounded from my reference speakers, the B&W Matrix 801 Series 3's. The Ohms' imaging, soundstaging, and spaciousness were distinctly different. These differences were not quite as apparent when I was sitting in my usual listening position, on the center line between the speakers. But oh, what a difference when I stood up and walked back and forth in front of the speakers, or walked closer to or in between them! The Ohms always maintained a more stable center image as I moved across the room. Even when I sat close up and directly in front of a system, the opposite speaker could be heard clearly; when I tried this with the B&W 801's, only the nearer speaker could be heard.

On most program material, the Ohms created a wide and stable soundstage, very realistic and spacious, with images that extended way behind the rear wall. With the B&Ws playing, I was very aware that the sound was coming from the speakers; with the Ohms playing, the sound was more diffuse and often seemed detached from the speakers. On material recorded in a large space with significant reverberation (such as choral, orchestral, and pipe-organ music), the Ohms' added spaciousness and realism were stunning. On other material, such as dryly recorded female vocals, I preferred the B&Ws. I almost always preferred the Ohms' sound on percussion instruments such as cymbals, drum rim shots, wood blocks, bells, etc.; the realism always went up a notch when I switched to the Ohms. I also preferred the Ohms on well-recorded chamber music, where the systems' added spaciousness increased the realism significantly. Often, when I switched from the Walsh 200 Mk-2's to the 801's, the sound field would collapse into the speakers, causing a significant loss of realism.

As an experiment, I tried re-aiming the Walshes, turning them 45° outward (facing their inside corners straight into the room) so that their axes of maximum radiation would directly face die listening area rather than crossing in front of it. This essentially negated the evenness of the Walshes' side-to-side coverage; but it hardly affected their spaciousness, because the speakers still had significant sound radiation to the sides and rear, which increased room reflections.

The Walsh 200 Mk-2's sometimes added an upper-bass chestiness to both male and female vocals, although otherwise the realism added by the Ohms on vocals was very compelling and appealing. Imaging and spaciousness aside, however, the B&W systems usually sounded more neutral than the Ohms. The Walsh Mk-2's also sounded somewhat distant at times, although this had more to do with their frequency response than with the spaciousness or reverberation that they added. On the same program material, the B&Ws had a more up-front, in-your-face sound. When reproducing pink noise, the Ohms had a distinct sound of their own, not objectionable but less smooth and more tonal than the B&Ws'. The Ohms handily passed the pink-noise stand-up/sit-down test, with barely noticeable spectral change.

The Ohm and B&W speakers had almost identical sensitivity, so I did not need to raise or attenuate the signal level to the Ohms when comparing them to the B&W systems. In the bass, the Ohms competed on a nearly equal footing with the B&Ws, even on loud rock music with heavy kick-drum and low organ-pedal notes. However, the B&Ws' bass was still somewhat tighter, a bit better controlled, and extended slightly lower than the Ohms'. Moving the Ohms closer to the corners elevated the bass level but did not improve the control. The Mk-2's could be played very loudly and cleanly on material that profits from high playback level, such as rock 'n' roll, country, and party music. But they also did justice to more sedate music because of their spaciousness and realism.

In Summary

The Ohm Walsh 200 Mk-2's have some extremely uncommon capabilities. Their nearly 360° sound radiation pattern below 1 kHz, and the way this maximizes room reflections, yields a strikingly realistic soundfield that extends across and between the speakers, and even behind them. This is both an advantage and a disadvantage, because it potentially makes the systems more dependent on room acoustics. The speakers' tailored radiation pattern provides a stable center image and soundstage for listeners located almost anywhere in the listening area, with no need to be equidistant from the loudspeakers. The Ohms add a large degree of spaciousness and airiness to anything that is played through them. Recordings intended to sound dry however, may not be so dry when played through these speakers. And the Ohms' robust bass capability should appeal to even pipe-organ aficionados. The Walsh 200 Mk-2 speakers can also be played loud and clean.

Do I like them? Yes! But before you run out and buy a pair you should definitely listen to them, to decide if their distinctive sound and uncommon capabilities suit your expectations and desires. Ohm makes this easy by offering a full money-back 120-day home trial, so you can try them out in your own listening setup for up to two months, then return them for full credit if you are displeased. For me, the Ohm Walsh 200 Mk-2's' soundfield-enhancing capabilities outweighed such minor problems as moderate tonal imbalances and the addition of spaciousness to material intended to sound dry.

—Don Keele

2-Way Minimonitor

B&W Nautilus 805

B&W Loudspeakers of America, 54 Concord Street, North Reading, MA 01864-2699. Voice: (978) 664-2870. Fax: (978) 664-4109. E-mail: marketing@bwaudio.com. Web: www.bwspeakers.com. Nautilus 805 minimonitor, $2000.00 the pair. FS-N805 stands, $600.00. Tested samples on loan from manufacturer.

The Nautilus 805, a beautifully crafted and elegant British miniloudspeaker, is less exotic-looking than some other speakers in B&W's Nautilus 800 series, but it's hardly a conventional box. Its cabinet is all curves, with hardly a parallel surface in sight; its only flat surfaces, in fact, are its front baffle and its bottom. Even in a head-on front view, it's saved from conventionality by the curvy transmission-line tweeter enclosure atop the cabinet. This visually arresting design simultaneously addresses the issues of diffraction, standing waves, and rigidity.

The tweeter in that top enclosure is B&W's 1-inch aluminum dome. The bass and midrange are handled by a 6.5-inch Kevlar driver, similar to the midrange driver in the Nautilus 801, but with a rolled rubber surround to give it more excursion and extend the bass response. The large port below the bass/midrange driver has B&W's characteristic dimpled surface, said to reduce port turbulence. The deep red cherry pair we auditioned had top-notch fit and finish.

The matching FS-N805 stands, made of extruded aluminum, add an eye-popping $600 to the Nautilus 805's price of $2000 per pair, but they and the speakers form a beautifully integrated pairing. The stands, which are available in black or aluminum finish, mount securely to metal plates on the speakers' undersides.

Measurements

The 805's performed quite well in the tests I was able to run (in-room sweeps with $^1/_6$- and $^1/_{12}$-octave resolution and various smoothing time windows). There was still adequate bass down to just below 55 Hz, and the speakers were quite flat in the midbass (where the response of many mini-monitors is tipped up) but delivered a bit of excess energy in the 200 Hz range. In my listening room, I measured a small trough of about 4 dB in the range from 800 to 1,000 Hz, but on many recordings that's a boon; it also has been shown that a dip in this region seems to add more sense of depth, something the 805 exhibited wonderfully. The step test (response to a DC input pulse) revealed very good time-alignment between the drivers, although the third-order crossovers do not permit true coherence.

I set the Nautilus 805s up in my smaller listening room (about 2000 cubic feet), placing them four feet out from the front wall. This arrangement gave me decent imaging and flattest overall response, though it somewhat reduced bass extension and impact—a common trade-off in speaker placement.

The 805s reproduction of music and voice was clean and wonderfully smooth. The treble reproduction was clean and without audible grain. With a pink-noise test signal, the Nautilus 805 proved to have excellent horizontal and adequate vertical dispersion. As with many speakers, response off the vertical axis dipped at the crossover frequency (about 3 kHz), but seated listeners will be on the vertical axis.

The 805 slightly editorialized the sound, which I attribute to a slight lower midrange excess and a dip in the upper midrange that I measured with both speakers working together in my room; I did not find the speaker to be as transparent in the midrange as it was in the treble, being a bit too polite (British?) for my taste in comparison to a pair of Paradigm Reference Studio/20's I had on hand. In the treble, however, the Nautilus 805 definitely outclassed the Paradigms, which tended to have a bit more bite on horns than the horns themselves did.

For most of my listening, I used the speakers full-range, driven either by one amplifier or by two separate amps, one feeding the woofer and the other the tweeter terminals of the speaker's crossover (passive biamp mode). In my 2000-cubic-foot room, they could play loud enough to satisfy me. On occasion, I did hear some glare in the midrange on difficult orchestral material, but only at levels a bit above what I consider to be home-concert level.

I also tried using the Nautilus 805's with various subwoofers and cross overs. My best results were with a Velodyne HGS-10 subwoofer and a Bryston 10B-sub crossover, the Bryston's high- and low-pass Butterworth filters set at 80 Hz with an 18-dB/octave slope. This combination let me take advantage of the 805's strengths, while getting low-distortion bass down into the twenties, and improved the dynamics by increasing headroom (generally 4 dB, according to speaker designers I've spoken to). This setup gave me flat, deep bass despite my having placed the B&Ws out from the wall for optimum imaging and flattest response. On the other hand, adding the Velodyne sub brings the setup's price to more than $5000, a price at which there are plenty of full-range speakers with comparable performance. So this combination is more of academic than of practical interest, unless you plan to use the Nautilus 805's in a den or as part of a home theater, where there's too little space for big, full-range speakers.

The Audio Critics longstanding policy for testing speakers is to combine objective measurements with several listening evaluations, preferably by at least two experienced listeners in at least two different rooms. So the B&Ws next went to the magazine's laboratory, for a full measurement workup and listening tests in another, larger room.

The laboratory's measurements were taken on a single speaker, quasi-anechoically (to factor out room effects). On-axis frequency response was pretty flat up to about 2.5 kHz, but above 3 kHz it was pretty consistently elevated: 2 to 3.5 dB above the 1 kHz output, rising to about +4 dB at 20 kHz. "Too much," said Peter Aczel. "They goosed the on-axis response to get flat power response into the room."

Sure enough, response 45° to the side was extremely flat. With the conventionally prescribed equilateral triangle listening setup, listeners would be 30° off axis, where response should show some of the elevated treble seen on axis. These are definitely not speakers to toe in so they directly face you.

At 45° above axis, the crossover dip extended from 1.5 to 5 kHz, but from about 5.5 to about 12 kHz the treble was still up by 2 to 4 dB, dipping again to reach a low of about -6 dB at 16 kHz, then starting to rise again.

In the bass, the measurements showed response rolling off to -3 dB at 40 Hz, a little better than B&Ws specified -6 dB at 42 Hz. Bass response was smooth and free of peaks, pretty close to the classic fourth-order Butterworth response, with a tuning frequency of about 37 Hz.

The Nautilus 805's impedance does not fall dangerously low; it reaches a minimum of 4.7 ohms at 200 Hz, and its magnitude from 500 Hz to 8 kHz is high enough to qualify it as an 8-ohm speaker overall. But the impedance phase takes some hefty swings, ranging from -57° around 90 Hz to +40° at 700 Hz. Most amps can handle that, but it would make some marginal amplifiers uncomfortable.

Distortion was reasonably, but not spectacularly, low. At a 1-meter SPL of 90 dB,. a 100 Hz tone produced a bit over 0.7% at the second harmonic and about 0.3% at the third harmonic. With the level raised to 100 dB SPL, the speaker's THD + N between 170 and 500 Hz averaged 1.5%, rising to 6% for frequencies below 100 Hz. That's pretty normal performance for a minimonitor.

Next, we auditioned the Nautilus 805's in ***The Audio Critics*** large listening room, which is less well damped than my room and whose listening position is twice as distant. Initially, with the B&Ws 3 feet out from the back wall, classical selections sounded ragged and unfocused. I suggested we move the B&Ws closer together, changing the listening setup from an equilateral to an isosceles triangle. This improved things considerably, but neither Peter Aczel nor I were enthralled, and I commented that this performance was much less satisfying than it had been in my quarters. The 805's sounded dynamically compressed on operatic recordings. The midrange sounded a bit ragged,

Manufacturer's Specifications

Type: 2-way vented-box system

Drive Units: 1" (25 mm) metal-dome tweeter and 6.5" (165 mm) woven Kevlar cone bass/midrange

Crossover Frequency: 3 kHz

Rated Frequency Response: 56 Hz to 20 kHz, ± 2 dB on reference axis

Frequency Range: –6 dB at 42 Hz and 30 kHz, –3 dB at 49 Hz and 22 kHz

Sensitivity: 88 dB SPL (2.83 V, 1 m)

Nominal Impedance: 8 Ω (minimum 4.6 Ω)

Power Handling: 50–120 W into 8 Ω on unclipped signals

Dispersion (within 2dB of response on reference axis): horizontal, greater than 60°; vertical, greater than 10°.

Harmonic Distortion (2nd and 3rd harmonics , 90 dB, 1 m): <1.0% 90Hz–20kHz; <0.5% 200Hz–18kHz

Dimensions: 16.3" (height) x 9.4" (width) x 13.5" (depth) = 415 mm x 238 mm x 344 mm

Weight: 20 lbs. (9 kg) each

Cabinet Finish: Real wood veneers–black ash, cherry, or red-stained cherry

Grille: Black cloth

and the recording's dimensionality was reduced.

Peter suggested we compare the Nautilus to our previous minimonitor champion, the Joseph Audio RM7si Signature, which is similar in price and size to the B&W. It was no comparison: the Josephs sounded much more detailed, transparent, refined, and dynamic than the 805's. To me, it sounded as though the B&W's bass/midrange driver were being taxed beyond its linear operating region, although even with smaller signals it lacked the transparency of the best minimontors we've heard.

Summary

All in all, I would call the Nautilus 805 a qualified success. It is beautiful in design and construction, carefully engineered, and without significant measurable vices. It can deliver fine sound with small signals, but others in its class can deliver such sound at higher volumes.

Perhaps in recognition of the Nautilus 805's limitations, B&W has announced a new version, the Signature 805. It has improved drivers, crossover, and bracing, as well as new finishes. Certainly, the driver and crossover improvements may well improve performance in the areas we found challenged in the original, and we would welcome a chance to assess the new version. At $1750 each, it is, alas, significantly more expensive.

The Nautilus 805 is recommended, but with qualifications due to its dynamic limitations and lack of ultimate transparency. It may well be your cup of tea; it just wasn't ours.

Noise-Canceling Headphones

Bose QuietComfort 2 Headphones

Bose Corporation, The Mountain, Framingham, MA 01701. Voice: (800) 999-2673. Fax: (508) 820-3465. Web: www.bose.com. QuietComfort 2, $299.00. Tested sample on loan from manufacturer.

Music shouldn't have to compete with noise but it always does. In our homes, where it's reasonably quiet, the competition isn't too fierce. But music is so portable these days that we take it with us to noisy places such as airliners. If we want to hear the music, we'd better quash the noise.

Bose's original solution to this problem, the QuietComfort headphones, made a good impression on me from the first flight I took with them. They fit comfortably, sounded good, and did a terrific job of keeping ambient noise from competing with the music they were reproducing. Bose's new model, the QuietComfort 2, does all that a little better and a lot more conveniently.

Unlike the earlier model (still available, at $249.00), the QuietComfort 2 folds flat to fit more easily in your carry-on, has a single cord instead of an easily tangled "Y" cable (an anachronism these days), and has its battery and circuits built into the earcups, not in a separate little box that weighs the cord down. Those are the main, but not the only, differences between the models.

Bose was the first company I know of to address the fact that high- and low-frequency noise pose separate problems, and provide separate solutions for each.

For the low frequencies, Bose employs active cancellation: Using built-in microphones to pick up the noise, the phones invert the noise signal's polarity and feed it to the transducers in its earcups. Within the earcups, the inverted signal cancels out most of the low-frequency noise that leaks in from outside. For higher frequencies, however, cancellation is not practical, because of the shorter wavelengths and, perhaps, higher processing speed involved. (In an airliner, headphones that reduce only low-frequency engine noise merely make it easier to hear annoying conversations in other rows.)

But high frequencies are easier to block than low frequencies. To block them, both the old and new Bose's earcups have hard shells and nonporous cushions that form a good seal against

(continued on page 52)

By Peter Aczel, Editor
David A. Rich, Ph.D., Technical Editor

High Efficiency Meets Hi-Fi, in Analog and Digital Embodiments

2- & 5-Channel Power Amplifiers

AudioControl Avalon & Pantages

AudioControl, a division of Electronic Engineering & Manufacturing, Inc., 22410 70th Avenue West, Mountlake Terrace, WA 98043. Voice: (425) 775-8461. Fax: (425) 778-3166. E-mail: info@audiocontrol.com. Web: www.audiocontrol.com. Avalon high-definition 2-channel home-theatre power amplifier, $1900.00. Pantages high-definition 5-channel home-theatre power amplifier, $2800.00. Tested samples on loan from manufacturer.

AudioControl has always represented no-nonsense engineering and solid value, untainted by either "tweako" cultism or el cheapo mass-marketing—our kind of manufacturer. (Their reasonably priced 1/3-octave real-time spectrum analyzer, reviewed in Issue No. 26, is just one example.) The two amplifiers under review here appear to be identical, except for (1) the number of channels and (2) the beefier power supply and fatter chassis of the 5-channel model. For that reason, I only tested a couple of channels out of the available 7, under the reasonable assumption that my measurements and conclusions will apply to both models (and therefore all 7 channels) equally.

The two amplifiers are well built; they even possess a certain degree of cosmetic polish, such as we are accustomed to from AudioControl. But of course they are totally lacking in high-end affectations such as half-inch thick sculptured front panels and fancy carrying handles.

The amplifier operates in Class H; this is a somewhat unusual configuration, based on tiered voltage rails

AudioControl Avalon

Features & Specifications

Type: 2- and 5-channel power amplifiers

200 watts per channel

High-definition BiMOS outputs

Cool-running Class H operation

"Light Drive" anti-clipping protection system

High-current gold output terminals

Output status display

Unbalanced RCA and balanced XLR audio inputs

Remote 12-volt trigger

Minimum speaker load: 4 Ω

Damping factor: 250+

Input sensitivity: 1.42 Vrms for full output

Input impedance: 30 kΩ

Dimensions
Avalon: 3½" (height) × 17" (width) × 15¾" (depth) = 89 mm × 432 mm × 400 mm
Pantages: 5¼" (height) × 17" (width) × 15¾" (depth) = 133 mm × 432 mm × 400 mm

Weight
Avalon: 22 lbs. (10 kg)
Pantages: 46 lbs. (20.9 kg)

AudioControl Pantages

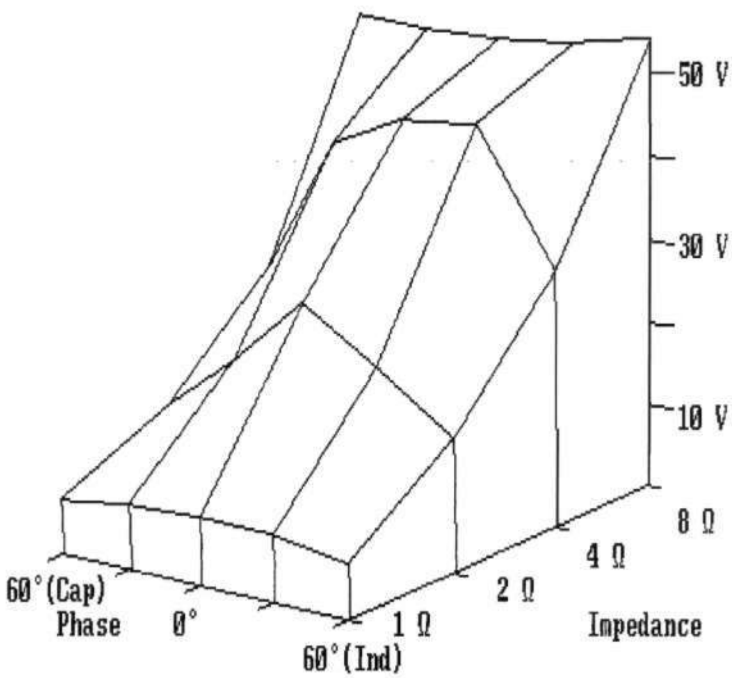
that permit low current draw with low-level signals and instantaneous high current draw with high-level signals. This lets the amplifier idle most of the time without dissipating large amounts of energy and allows cool operation with thermal convection only, not to mention fairly compact size. (A more sophisticated implementation of the same concept is Bob Carver's "tracking downconverter" as used in his Sunfire amplifiers.) The output devices are BiMOS power transistors, claimed to combine the best characteristics of bipolar and CMOS technologies. (I have no supporting data.) A nice feature is the availability of both unbalanced (RCA) and balanced (XLR) audio inputs. The various output status lights are consolidated in a handsome large window. All in all, it's a pretty slick design.

Measurements

The most basic measurement of any power amplifier is distortion versus output power at various frequencies. The channels I tested were virtually identical—and not particularly impressive, distortionwise. Into a load of 8Ω, clipping occurred just above 200 watts, but this was not the point of minimum distortion as is the case where the distortion is completely noise-dominated. With a 1 kHz input, the distortion curve bottomed out at 16 watts and again at 67 watts, at which points it was in the -81 to -82.5 dB range. At clipping, distortion rose to -77.5 dB. With a 20 Hz input, the minimum was at -85 dB from 30 to 60 watts. The 20 kHz distortion, however, was about 10 to 20 dB worse (-73 to -63 dB) at power levels above 10 watts, indicating rather severe dynamic distortion. Into a 4Ω load, clipping occurred at 340 watts, with the 1 kHz and 20 Hz distortion curves retaining much the same profile at proportionately higher power levels and the 20 kHz distortion curve flattening out at -62 dB right up to 300+ watts.

Now -62 dB is only 0.08%, so you could say "what's the big deal?", but the fact is that cheaper amplifiers with conventional power supplies often have considerably lower dynamic distortion. The —80 to -85 dB minima for distortion at the lower frequencies are far from bad, but in a classic circuit they would be located at clipping, not at lower power levels.

Probably more revealing of the true nature of the Class H circuitry is our PowerCube test (short-burst power into 20 different resistive and reactive loads), which no other American audio magazine performs. This



PowerCube performance of AudioControl amplifier, showing short-burst output at 1 % distortion for 20 different loads. The center (0°) line shows performance into purely resistive loads; the others show performance into increasingly reactive loads, capacitive loads to the left and reactive loads to the right. The vertical scale shows output voltage. Output into 8-ohm loads is just as it should be, but power slopes off rapidly at lower impedances, a sign of power-supply limitations. The power dropoff with reactive loads of 4 ohms or less shows that such loads trigger the amplifier's current limiting. See text.

test clearly showed that the AudioControl amplifiers are really happy only with 8Ω loads, which drew outputs of 365/350/343/344/361 watts into -60°/-30°/0°/30°/60° impedances at 1% distortion, i.e., slightly more into reactive than resistive 8Ω loads—the proper response. With 4Ω loads the picture changed radically, the output being far greater into 0° than into the reactive loads. Lowering the loads to 2Ω greatly exaggerated this anomaly and when it came to 1Ω loads—forget about it, hardly any outputs at all. Give the amplifiers credit, however, for not shutting down with any of the 20 different loads, as some do.

Frequency response at 1 watt into 8Ω dropped to -0.1 dB at 10 kHz and -0.34 dB at 20 kHz, a somewhat steeper high-frequency rolloff than the norm. At 20 Hz, the response was down to only-0.08 dB. Crosstalk between two adjacent channels was 10 to 13 dB better in one channel than the other: -57 and -67 dB at 10 kHz, -72.5 and -86 dB at 1 kHz, converging to -93 dB at 40 Hz. Those are all pretty good figures. Signal-to-noise ratio at clipping was between 95 and 97.5 dB in the channels I measured—also good.

Does all of the above add up to a recommendation? Maybe a qualified one. These units aren't exactly cheap, but they are physically attractive and compact packages, and into 8Ω loads their performance is basically flawless. If you have speakers whose impedance tends to dip low and turn highly reactive at various frequencies, then there exist better choices in amplifiers. Since the majority of speakers have a nominal impedance of 8Ω, the AudioControl amplifiers can certainly be recommended to drive them. A more sweeping endorsement isn't warranted, despite our respect for AudioControl's engineering.

—*Peter Aczel*

DVD Audio/Video Player & 7-Channel AV Surround Receiver

Denon DVD-9000 & AVR-5803

Denon Electronics (USA), Inc., 19 Chapin Road, Building C, Pine Brook, NJ 07058-9777. Voice: (973) 396-0810. Fax: (973) 396-7448. Web: www.denon.com. DVD-9000 DVD-Audio/Video player, $3500.00. AVR-5803 AV surround receiver, $4300.00. Tested samples on loan from manufacturer.

Let us note right up front that the Denon AVR-5803 receiver is identical to the superb AVR-5800 reviewed in Issue No. 27, except for a limited number of new features, the most important of which is the Denon Link terminal. Using a special cable, the Denon DVD player can be digitally connected to the Denon Link terminal on the updated receiver, thereby avoiding redundant D/A and A/D conversions in multichannel playback and allowing multichannel bass management. That is the main reason for bracketing the two units in the same review. Other new features of the AVR-5803 include DTS 96/24 compatibility, Dolby Pro Logic II decoding, and Dolby headphone compatibility, none of which can be quantified with standard measurements; they can only be evaluated subjectively. We are therefore not reviewing the updated receiver all over again, nor do we list again its many features, except to add some general commentary further below to the previously published review.

The Denon Link would not be necessary if the S/PDIF and optical connections were allowed to transmit digitally encoded multichannel sound, but the industry has prohibited that approach out of fear of a too-easy/too-perfect copying capability. (It's a complicated and highly controversial scene, deserving a separate article.) The Denon Link is a proprietary workaround, based on an encryption process, to obtain a digital multichannel connection. Let us be grateful, under the circumstances, that it exists at all.

Quite aside from the Denon Link feature, the DVD-9000 is a handsome, surprisingly massive (41 lbs.), and highly capable piece of equipment—and at $3500 it had better be. Indeed, at that price, I expected it to play SACDs in addition to DVD-Audio/Video discs and standard CDs, but it doesn't. Too damn bad. What it does, however, it does just about perfectly.

Whatever test discs I have on hand

Denon AVR-5803

Features

DVD-9000

Denon Link (digital connection to AV receiver)

Newly developed AL24 Processing Plus

High-precision multi-24-bit D/A converter

HDCD decoder

Dolby Digital (AC-3)/DTS decoder

THX Ultra certification

Progressive-scan circuit ("Pure Progressive")

14-bit/108-MHz video D/A converter

Vibration-resistant design

Usable as D/A converter

Multiple language/subtitle/angle/GUI functions

AVR-5803

See AVR-5800 features in Issue No. 27

Denon DVD-9000

were reproduced through the analog outputs of the DVD-9000 within a dB or two of theoretical perfection. Only a couple of high-end Sony players I have tested in the past were its equal in that respect, exhibiting no gain-related analog distortion at full scale (0 dB), among other things. I won't even specify the exact numbers, however, because they are basically irrelevant. Nobody in his right mind would spend $3500 on the Denon because of its great performance through the analog outputs. Not in the era of perfectly good $200 DVD players. Its only raison d'être is the straight-through digital playback of multichannel recordings in conjunction with the AVR-5803. That's a $7800 package, but at least no one can say that the same thing is available for a fraction of the price. (Actually, the AVR-5803 is such an exceptional AV receiver that, in its case, I am almost resigned to the astronomical retail price tag.) Don't misunderstand me. I am not saying that the DVD-9000 isn't "better" than a $200 player. Of course it is. But the difference is in general solidity, smoothness of operation, a few extra features, subtle video detail, etc., etc.—it's not a machine from another planet as the price difference might indicate.

A few features of the DVD-9000 are worth noting. The remote control is nicely laid out and relatively easy to use, with discrete up/down/left/right buttons instead of the annoying joystick control of so many other units. The "newly developed AL24 Processing Plus" is hyped by Denon as "an advanced version of conventional AL24 Processing." It used to be called Alpha System Processing and is a nebulous, very sketchily explained signal compensation technique that David Rich dismissed in Issue No. 22 as "designed to make undithered sine waves look good at hi-fi show demos." There may be some vague benefits, but I'll be damned if I can figure them out, and Denon's mysterious statements are no help. HDCD is a rarely used encoding/decoding technology, transparently compatible with conventional digital recordings, whose main benefit is increased dynamic range (at least in theory). I haven't seen too many HDCD-encoded recordings lately (Reference Recordings is one label that has consistently stuck with it), but it's only right that a $3500 player possesses the decoding capability. As for progressive scan, it is undoubtedly an important advancement in video resolution, but it requires a compatible TV monitor.

When it comes to the AVR-5803 receiver, it was my opinion (see Issue No. 27) that its almost identical predecessor, the AVR-5800, represented the state of the art in AV surround receivers, so the "new improved" version automatically advances to that rank. Among its new features the most important and impressive by far (other than the highly specialized Denon Link) is the Dolby Pro Logic II decoding capability. For the first time, there can be no argument about playing even conventional stereo recordings, not just 5.1-encoded program material, in the 5.1 mode. It's a considerable improvement over the earlier Dolby Pro Logic. The sound field is more convincing, beginning to approach discrete 5.1-channel reproduction in quality, even with just plain stereo CDs. As far as video performance is concerned, I noticed no significant differences from the AVR-5800.

There remains the Denon Link to be subjectively evaluated, and that's a problem. Today's best D/A and A/D converters are so perfect—and Denon uses only the best in their high-end equipment—that leaving them out of the loop appears to make little or no audible difference. My exposure to the DVD-9000-cum-AVR-5803 combination has been relatively brief; I would really need more time and a greater variety of multichannel program material to fine-tune my aural perception of the difference between the two modes—if any. That doesn't alter the fact that bypassing the D/A-to-A/D conversion process is absolutely the right thing. It makes sense. It's the way things should have been done since the beginning. That bass management is now possible with DVD-Audio signals, which it wasn't through the six external analog connections, is alone a considerable advantage. Unfortunately, the Denon Link does not work with copy-protected discs. Those discs don't know the difference between a proprietary and a generic digital connection. Don't blame Denon, however, for the idiocy of the recording industry.

—Peter Aczel

CD/SACD/DVD Digital Disc Player/Receiver

Sony AVD-S50ES

Sony Electronics, Inc., One Sony Drive, Park Ridge, NJ 07656. Voice: (201) 930-1000. Fax: (201) 358-4060. Web: www.sony.com. Model AVD-S50ES multichannel player/receiver, $500.00. Tested sample on loan from manufacturer.

There are plenty of good reasons to combine a player for CDs, DVDs, and Super Audio CDs (SACDs) with an audio/video (A/V) receiver, as Sony has done in the AVD-S50ES. For starters, there are the usual advantages of combining components: savings in cost and shelf space, plus simpler operation (e.g., the receiver automatically knows when a disc is playing). And with a multichannel setup capable of playing SACDs, these advantages are even more pronounced; for example, the Sony's direct internal digital connection from player to receiver eliminates the need for as many as eleven cables (for 5.1 analog channels, digital audio from CDs and conventional DVDs, and both S-Video and component video). Furthermore, while SACD players usually have digital outputs for CD and DVD playback, they have only analog outputs for SACD, to prevent unauthorized digital copying. So a multichannel SACD player will need six digital-to-analog (D/A) converters, raising the player's cost. Putting an SACD player and receiver in one box provides a secure digital interface between the two, saving the expense of added converters.

Because the multichannel analog inputs on A/V receivers and controllers commonly bypass all processing stages except the volume controls, the bass-management and speaker-distance-compensation functions built into such systems aren't available for SACD playback. Including these systems on an SACD player would further raise its cost. In theory, a player-receiver combination like this Sony permits implementation of such features. In the AVD-S50ES, however, the SACD signal enters the digital power amplifier after the digital signal-processing (DSP) circuits that do this compensation; thus this product does not appear to have this capability. Unfortunately, my review showed that this was but one of the capabilities the AVD-S50ES lacked, despite its advanced technology in other areas.

Technical Design

To squeeze a multiformat digital disc player and a receiver into a box that is smaller than a conventional A/V receiver alone, you must do something radical to improve the efficiency of the power amps. Sony has chosen to use its S-Master digital power-amp design, which is based on its one-bit, delta-sigma modulation technology. In the S-Master design, all signal processing within the unit takes place in the digital domain—but in pulse-code modulation (PCM), not one-bit, form. At the power amp stage, a DSP chip converts the PCM digital signal to a one-

bit digital signal (but with a different sampling rate and noise shaper than SACD). Next, an intermediate IC scales the one-bit signal's voltage up from 3 volts to a signal large enough to drive high-speed power MOSFETs. The power MOSFETs switch between the two supply rails, thus forming a high-voltage one-bit signal. The radio-frequency energy of the one-bit signal present at the power MOSFETs' output is removed by passive filter circuits, and the filtered signal is sent directly to the speaker terminals. I predict that this is how almost all mass-market A/V units will work in the future. However, as we'll explore later, audiophiles won't find this approach optimal now, and possibly not ever.

In discussing the design of its digital power amp, Sony's press release states that "this one-bit signal can, in effect, be turned into an analog signal simply by filtering out the digital sampling frequency, so that the signal appearing at the speaker terminal is essentially the digital signal itself." However, as Sony itself admits, it's not that simple. For one thing, SACD signals contain much out-of-band noise, due to the aggressive, fifth-order, noise-shaping used in the SACD system. This is no problem with standalone SACD players, whose low-voltage output signals are normally carried via shielded cables. But such signals won't do as output to the speaker terminals, where signal amplitudes are higher and the unshielded cables used could act as antennas, broadcasting the noise. Since switching voltages between two levels in the Sony design generate out-of-band signal components, the noise must be filtered out.

In the AVD-S50ES, this filtering happens in the amplifier output stage, where voltages are comparatively high and currents fairly large (at the Sony's rated power—100 watts per channel into 6 ohms—output amplitude is about 25 volts rms, with about 4 rms amperes of current); this calls for passive filter components capable of handling high voltages and currents. Furthermore, the rise and fall times of the digital signals at the speaker terminals, and the time uncertainty as to when the switching occurs (capture jitter), create opportunities for noise and distortion to arise.

The S-Master design (see sidebar, "Circuitry of the Sony AVD-S50ES")is an all-digital implementation of a switching amplifier. This is the Holy Grail of switching-amplifier design, with analog signals present only at the speaker terminals. While pure digital solutions are the most elegant implementations, it is generally agreed that switching amplifiers of ultimate quality will require an analog feedback loop around the digital section, to reduce the distortion introduced by the nonideal waveforms the MOSFETs produce. (The issues are the timing of the switch's turning on and off and the shape of the waveform as it rises and falls).

Listening Tests

I'm an audiophile, not a videophile, so to me the AVD-S50ES is not a DVD player that can also handle SACDs, but rather a device for playing music in surround (the best way to hear it) that also happens to play DVD movies. The Sony can handle several forms of surround: discrete, uncompressed multichannel sound from SACDs; discrete surround, with some data reduction, from Dolby Digital or DTS soundtracks on DVDs; decoded matrix surround from videotapes and other two-channel sources carrying Dolby Surround soundtracks; and simulated surround generated from stereo CDs and other sources.

My listening setup for this review was optimized for audio; I feel that a mixed-use audio/video system entails too many compromises. The front left, center, and right speakers were matched AR 302's. The rear channels were Monitor Audio Studio 6 speakers, which have enough bass to be classified as "large" in surround setups. All the speakers were equidistant from my ears and mounted in the same vertical plane. Monopole speakers like mine should also be several feet from any wall (and were), though this is less critical for surround speakers.

Like other surround components, the AVD-S50ES has adjustable delays to compensate for unequal speaker distances (except when playing SACDs; these adjustments are bypassed in that mode). However, moving the center

Features

- Combination receiver and digital disc player
- Plays DVD, CD, SACD, CD-RW, and MP3 recordings
- All-digital amplification
- 5-channel amplifier
- 100 watts per channel (6 ohms, 20 Hz–20 kHz, 0.7% THD)
- Dolby Digital, Dolby Pro Logic II, Dolby Pro Logic, and DTS decoding, plus eight Cinema Studio and other soundfield simulations
- Preprogrammed A/V Remote Commander remote control
- FM/AM tuner with 20 FM and 10 AM station presets
- 1-bit audio D/A converter
- Component video and S-video outputs
- Interlaced and progressive-scan (480p) video outputs
- 10-bit video D/A converter
- 3 analog audio inputs, 1 output
- 2 optical and 1 coaxial digital inputs, 1 optical digital output
- 3 S-video inputs, 2 outputs
- Subwoofer line output
- Headphone jack

speaker relative to the left and right ones reduced the realism of the sonic picture, as did moving the left and right front speakers closer to the wall behind them, even when the distance compensation was adjusted for this. (Distance compensation was more effective when moving the rear channels closer in than it was for movement of the center channel.) Moving the center speaker farther from the listener than the left and right ones creates a cavity effect, with some of the center speaker's sound scattering from the sides and back of the other front-speaker cabinets. And when the center speaker is moved closer to the listener, it blocks or scatters sound from the left and right front speakers. (More problems arise when the speakers are used for home theater—see sidebar, "The Center-Channel Conflict.") Delay adjustments compensate only for arrival-time differences, not for acoustical effects that occur when speakers are placed too close to a back wall or are not in the same vertical plane.

The placement of the surround-channel speakers also proved important. Improper placement, with the rears not the same distance from the listener as the main speakers, caused some instruments to wander into the rear channels. Although the rears should be about the same distance from the listener as the front speakers, they should be placed farther apart, for a wider listening angle. The diagrams in the Sony manual and the Telarc CD booklets basically describe this setup, which is optimized for music listening and differs from the optimal layout for movies. Using Sony's and Telarc's recommended layout was no problem for me—I don't do movies. For almost everybody else, movie placement wins.

The AVD-S50ES was not only the first SACD player I've used, but also the first component I've had that incorporated Dolby Pro Logic II (PL II) surround decoding. The original Pro Logic delivers an okay sort of surround from stereo recordings, even though it was not designed for that. PL II was designed with that use in mind, so naturally it does a better job of converting stereo to surround than the older system; it even has separate Movie and Music modes. But, to my surprise, the AVD-S50ES's PL II decoder did a better job of converting stereo music to surround in Movie mode than in Music mode.

On systems having left, center, and right front speakers, PL II Movie can transform a good orchestral stereo recording remarkably. (The surround speakers, in my opinion, make less of a difference.) With three speakers across the front, the center is defined, woodwinds gain definition, and the brass gains body; moreover, the woodwind and brass images are more towards the back of the sound stage, where they are supposed to be. Strings sound fuller and less bright, too, aided by both the center and ambience (surround) channels. And all instruments in a concert-type recording gain definition and detail, because ambience information is moved to the rear, making the front channels drier.

The problem with the original Pro Logic is its gain-steering circuit, which was designed to keep center information in the center at all costs. This is good for movie voice tracks, but not so good for vocal music, and one can sometimes hear the attack and decay as the steering logic operates. Dolby Pro Logic II has more advanced logic, which operates less noticeably and more naturally. Pro Logic II also enhances ambience by having stereo surround channels.

In A/B comparisons, I heard significant shifts in tonal balance when switching from Pro Logic to Pro Logic II Movie; with large-scale orchestral music, I preferred the PL II Movie mode to plain Pro Logic in almost all cases, but on chamber music, Movie mode reduced definition and made instruments sound larger than life. The alternate PL II mode, Music, sounded better than the Movie mode on some chamber music.

Pro Logic II Music should have been

The Center-Channel Conflict

In a home theater setup, there is usually a conflict between the center channel and the TV, arising from audio requirements. For optimal multichannel reproduction of music, the center speaker must match the frequency response of the left and right front speakers all the way down to the subwoofer cutoff. Also, the center speaker's distortion must be at least as low as that of the left and right speakers (preferably lower) because it often must work harder than they do.

Center speakers small enough to be placed in front of the TV without blocking it, or on top of the TV, don't meet these requirements; minimonitors, for example, frequently reach 10% distortion at 80 Hz with an output of 95 dB SPL. Even when the three front speakers are identical, a center speaker placed right in front of a TV, or above it, is operating in a different acoustical environment than the other two; this also alters its frequency response. These problems can be overcome: in *The Audio Critic's* lab, for example, the speaker setup is optimized for music and a front-projection screen is placed high on the wall, above the top of the center-channel speaker.

Circuitry of the

The details of the Sony's design and operation were not made available to us, nor was a service manual. Without a manual, I was left to literally poking around the unit.

Opening the AVD-S50ES reveals, Toto, that we aren't in Kansas anymore. In Sony's A/V digital Oz, the receiver's power supply is dominated by heat sinks for the power-hungry digital electronics, which run on standard, class A voltage regulators. The high-efficiency audio power amplifiers are powered by a high-efficiency switching power supply, whose biggest transformer is less than 2 inches on a side. Absent are massive transformers, filter caps, and output-transistor heat sinks-all of which would be highly noticeable in conventional 100-watt/channel receivers.

Most mass-market consumer electronics use relatively less expensive, single-sided boards. The AVD-S50ES uses double-sided boards because of the complex network of digital integrated circuits (ICs) they interconnect and because of Sony's desire to keep the unit compact. (Even the power-supply board is double-sided.) The boards are stacked on top of each other in some places, with ribbon cable running between them. Anything analog (such as S-video and audio switching) happens on the rear of the unit (or so it seems-the locations of the audio A/D and video D/A converters were not clear.) The whole signal-processing engine and the microcontrollers that coordinate all the functions of the receiver are on a single PC board of just 4½ by 7½ inches. One DSP chip is covered with a very large heat sink, a sign that it's crunching data at a high rate and dissipating a lot of heat.

A problem with class D amplifiers is that their power-supply rail voltages must stay absolutely constant, whereas the speaker load requires the amplifier to source or sink large amount of current. Unlike a class AB amplifier, a class D amplifier has no power-supply rejection; if the supply-rail voltages are not absolutely constant, distortion results. The small size of this unit's switching power supply makes me wonder if the power rails are kept constant enough. The PowerCube measurements given in the main text of this review point to potential problems here.

The power amp PC board, at the rear of the receiver, holds three 80-pin surface-mount chips, each connected to two additional small chips, which in turn connect to the power MOSFET devices. These devices are hidden beneath a metal plate, 2½ by 1½ inches, marked "S-Master" (this is the only heat sink in the signal circuitry). The plate is glued directly to the MOSFETs, making it impossible to find out what MOSFETs are used. The output from these secret MOSFETs goes off to a rather complex network of inductors and capacitors that forms the RFI filter for the S-Master output stage. This filter drives the speaker terminals. The S-Master output stage is balanced, with voltages on the positive and negative speaker terminals moving in opposite directions rather than one moving while the other serves as ground. It thus follows that the RFI filters must be used in pairs for each output channel. The components in the RFI filter are much smaller than one would expect in a 100-watt/channel amplifier, leading me to wonder about saturation of the inductors at high currents, which would be consistent with our THD measurements.

So the key to how the Sony does the audio D/A conversion and power amplification is in the LSI (large-scale integration) chips and the stuff under the S-Master plate. Questions to Sony yielded no answers, but a nifty Web site on class D amplifiers (www.classd.com) did. This site offers a wealth of information on the many class D amplifier ICs for audio applications. While some of the opinions on the site are questionable, and one must thus be careful before taking all the information as accurate, the overall site is extremely useful for anybody who is interested in the technology. (The site also presents sonic evaluations of the circuits, although these are not of any use because they are not based on double-blind listening.)

From the class D site, and the Mitsubishi IC site it links to, we learn that the power-amp section of the AVD-S50ES uses two types of LSI chips made for Sony's S-Master system by Mitsubishi. The first is an all-digital chip (one required for each two channels) that eventually converts the PCM or SACD digital data into the S-Master one-bit stream. The second set of chips (two per channel), are predrivers that convert the small S-Master digital signals into digital signals with much higher voltages and current capability to drive the digital power MOSFETs. The predriver is also involved in protecting the MOSFETs, including formatting the signals to these power devices so that the two FETs are never on at the same time.

The MOSFETs themselves are critical to the performance of the switching power amplifier. To prevent signal distortion, they must switch at very high speed and achieve very fast rise and fall times. They are especially critical in a design like this, which has no feedback, thus laying bare the distortion produced by the MOSFETs or by the RFI filters for all to see and maybe hear.

The data sheet for the Mitsubishi digital audio processor chip hands us a big disappointment: the SACD bitstream is converted into PCM and then re-modulated at a slower data rate (768 kHz vs. 2.28 MHz) by a delta-sigma modulator in the processor chip. The chip also accepts PCM data at a variety of sampling rates. To resample all the different input data rates (from CD, MD and DVD) to one sampling rate, the Mitsubishi chip uses an asynchronous sample-rate converter. With this circuit within the Mitsubishi chip, it is possible to use a clock that is independent of the clock associated with the incoming data stream. The Mitsubishi chip generates a low-jitter clock, using a crystal oscillator on the power-amp PC board; an external system clock from the DSP board would have an unacceptable amount of jitter for a digital power amp using an oversampled class D noise shaper. And in an all-digital switching amplifier that uses no feedback, jitter levels on the data output of the digital processor chip must be very low.

But if, as here, a new clock is generated on the power-amp board, we no longer have a synchronous system with one clock locking all the digital circuits together. To operate a PCM-based system on multiple clocks, an asynchronous sample-rate converter is mandatory; such a converter is part of the Mitsubishi S-Master digital chip.

The Mitsubishi chip handles volume-control attenuation in the digital domain, before the delta-sigma modulator. This requires the modulator and switching power amplifier to have extraordinary dynamic range. In a traditional A/V receiver, small volume changes (3 dB or less) are accomplished digitally but larger steps are handled in the analog domain. In such receivers, the analog volume-control function occurs between the D/A section and the analog power amp, ensuring that any noise generated in the D/A conversion process will be reduced as the volume is reduced; the D/A converter can thus get by with a smaller dynamic range than would be needed in an all-digital implementation.

The data sheet gives very limited information on the Mitsubishi digital amplifier chip's delta-sigma modulator. The modulation frequency is very high for a class D power amplifier (768 kHz-one fourth of SACD's top sampling frequency), requiring extraordinarily fast power MOSFETs. With the identity of these MOSFETs concealed by the S-Master plate, there is no way to ascertain their performance characteristics. One assumes the higher sampling rate was chosen in order to produce a class D amplifier that could accurately reproduce frequencies in the range from 20 to 40 kHz, which can be produced by SACDs. But as SACD's data rate is 4 times higher than the S-Master's, the amplifier's signal-to-noise ratio is lower than that achieved by SACD's Direct Stream Digital (DSD) data stream. In addition, the noise shaper in the S-Master system must be designed to ensure that the passive RFI filters at the amplifier outputs can reduce the out-of-band energy in the S-Master data stream to below FCC limits.

Noise shaping in the DSD data stream shifts noise energy outside the audio band in order to reduce noise within the band. This data stream therefore has very high levels of high-frequency energy, which carry through to the high-level analog signal and could therefore radiate from the unshielded speaker cables to other components in an audio system. To prevent this, the S-Master must use a different noise-shaping algorithm than DSD, to shift some of this noise energy back into the audio band.

It can thus be seen that the S-Master data stream and SACD's DSD data stream have very different characteristics, with the S-Master stream being unable to produce the DSD stream's low distortion and high signal-to-noise ratio. This situation is not improved by having the volume-control stage precede the S-Master modulator, where it discards some of the signal data when the control is turned down.

One is left to wonder how Sony overcame the normal problems associated with high sampling rates in switching power amps (768 kHz is about twice the normal rate for a switching amplifier of this type). One of these problems is an increase in switching losses: the class D output stage is only efficient when there is no voltage across the MOSFETs' drain and source terminals. But there is a voltage while the MOSFETs are switching between positive and negative rails. Doubling the clock rate of a class D amplifier doubles the number of edge transitions, and switching losses increase.

Another problem with faster switching is dead time (the time when both the pull-up and pull-down transistors are off). A certain amount of dead time is essential, lest the pull-up and pull-down transistors turn on together, short-circuiting one transistor to the other (in engineering land, we call this crowbar current). This dead time stays constant as the switching frequency increases, causing it to be a larger percentage of each switching cycle. Needless to say, you do not get efficiency when the circuit crowbars; but any dead time in the output stage results in distortion, since the output just floats during that time rather than being pulled to one of the desired states.

Finally the higher speed results in less accurate matching between the rise and fall time of the pulses. This, too, results in distortion.

Summing up the technical discussion, we find the technology Sony has chosen to be at the bleeding edge. I would have thought it impossible to create a consumer product with a 100-watt/channel all-digital class D amplifier, switching at close to 800 kHz, with full digital volume control, and a microscopic switching power supply to provide 500 watts of power to the five speakers. But Sony, with the LSI chips Mitsubishi made for it, has turned the apparently impossible into a commercial reality.

-David Rich

better for all music, but on the AVD-S50ES it usually was not: too much information moved from the front to the surround channels; the front soundstage was too wide; and individual instruments were placed far less accurately in the soundstage than was true in the Movie mode. These factors are controlled, respectively, by three parameters: Panorama, Center Width, and Dimension; they are found in PL II Music, but not Movie, mode. On many surround processors, these parameters can be set by the user, but I could find no way to do this on the AVD-S50ES. While I cannot be positive without some way of changing these settings, it appears to me that Sony switched on Panorama (which extends the front image out to the surround channels for what Sony calls "an exciting 'wraparound' effect"), set Center Width too low, and set Dimension (which moves the surround field forward and back within the room) too high. Conceivably, these settings are correct for some speaker setup other than mine, but it should be possible to adjust them for any reasonable sound system. I feel certain that, were I able to adjust these parameters, Pro Logic II Music would become the winner it should be.

Meanwhile, the Dolby Pro Logic II Movie mode is a music lover's dream, dramatically enhancing the sound of well-recorded discs. It is so good on most classical-music CDs—as good as true SACD discrete surround discs in many cases, and sometimes better!—that I'd never want to go back to stereo for orchestral music

I had four surround SACDs. Three were from Telarc: Rachmaninoff's Piano Concerto No. 3 with Chinese pianist Lang Lang and the St. Petersburg Philharmonic under Yuri Temirkanov, Sibelius's Symphony No. 2 with Paavo Järvi leading the Cincinnati Symphony Orchestra, and Vaughan Williams's *A Sea Symphony* with Robert Spano conducting the Atlanta Symphony Orchestra and Chorus. One was from Sony: Mozart's Sinfonia Concertante in E-flat Major, K. 364, and Concerto in D Major for Violin and Piano, K.Anh.56, with Midori on violin, Nobuko Imai on viola, and Christoph Eschenbach on piano and conducting the NDR-Sinfonieorchester. These are all dual-layer discs, with SACD data on one layer and CD-compatible stereo PCM data on the other. However, the AVD-S50ES will not play the CD layer of such discs; to hear each disc's stereo mix, I had to make a CD-R copy of its stereo PCM layer (which plays fine on regular CD players), then play that copy on the AVD-S50ES.

To my surprise, the SACD surround tracks on the three Telarc discs sounded strange compared to the same recordings' stereo mixes played in Pro Logic II Movie mode. The issue appears to be Telarc's unusual use (or, rather, almost nonuse) of the center channel. Almost nothing comes out of the center—some woodwinds, but limited strings and no brass. The result is a soundstage that confines stage-center instruments to the center channel and puts too much sound at the left and right sides of the stage. The Telarc Rachmaninoff recording is a crazy mix: the piano is spread from full left to full right, and only the piano—not the orchestra—appears in the center. The orchestra soundstage is pasted into the piano soundstage, with no relation between the two. With no center info, the orchestra does not sound as good as it does when the stereo version is played with Pro Logic II Movie decoding. (Furthermore, the piano sounded as it would if you were very close to the instrument and facing the keyboard, as if playing it—though it sounded very good once I accepted this perspective.)

The Sony disc (Mozart Double Concerto) sounded much better. The orchestra's sound was truly excellent, with great clarity yet with body and warmth. On orchestral passages, the center channel was used correctly, with lots of woodwind content but also with lower levels of strings and brass to better center them in the soundstage. The soundstage for the violin and viola solos, however, was completely messed up, with the center channel carrying mainly low-level echoes of the solos while the violin was placed hard right, the viola hard left. In real life they would be much more centered. I tried the old Marriner stereo recording of this piece (same viola player, in a better performance) in Dolby Pro Logic II Movie mode. The soloists were much more realistically placed, but the orchestra sounded less clean.

Usage Tests

To my great surprise, the Sony AVD-S50ES has only S-video and no composite-video signal inputs, so you cannot hook up the many VCRs or cable boxes that have only composite outputs. On the other hand, the component-video outputs worked okay; I even got my three-year-old, first-generation, Panasonic progressive-scan TV to lock to the AVD-50ES in that mode. Since I am not a videophile, I loaned the Sony to a friend who is. Three months later, he reports no problems when using the unit in home-theater mode. (In fact, he found the advantages of the single-box system so compelling in comparison to his big, complex A/V receiver and separate DVD player that he decided to purchase our test sample.)

The Sony's front-panel controls are very rudimentary—a pain, because you are therefore forced to work with the remote (if you have not misplaced it) to adjust and use the receiver—and the remote has only identical, microscopic buttons. On the good side, the front-panel display of the AVD-S50ES is informative enough that you do not need to watch an on-screen display to adjust and control the unit. Even with the remote, adjustment modes are minimal: no tone controls, no adjustable subwoofer crossover frequencies. And the

AVD-S50ES has one fewer surround-speaker placement mode than my $300 Sony A/V receiver. (That receiver has settings for side, mid, and rear surround-speaker placement, corresponding to angles of ±90°, ±60° and ±30° from the center speaker's axis; the AVD has only the side and rear settings.) Selecting modes and making settings on the AVD takes a lot more button-pressing than on my receiver, because it forces you to go through menus to get into the speaker-placement, speaker-level, and surround-algorithm selection modes; my receiver has dedicated buttons for these choices. The AVD also has no THX modes (no Sony units do) and, because it has no back-speaker outputs, does not support Dolby EX or Neo: 6 (the DTS equivalent).

There were also some performance issues. The AVD-S50ES takes a long time to spin up a disc, and the transition time from CD to SACD mode can take a full minute. On some classical CDs, the Sony appeared to clip off the first one or two tenths of a second of the first track. Loud pops occurred when I used the front-panel switch to select inputs, but not when I used the remote for this. (With one SACD I got almost continuous dropouts, and had to stop the player and reseat the disc to get it to work—that's probably the disc's fault, but I've had no chance to try it on another player.)

Dolby Pro Logic and Pro Logic II are disabled when a disc with discrete 5.1-channel sound is played. Unfortunately this also happens when a *stereo* SACD is played. And the player stays in stereo mode when you switch back to listening to CDs, even if you had been listening to them with Pro Logic previously; it should remember your settings rather than making you turn Dolby Logic back on again.

The output terminals are balanced and floating, with no fixed ground, so you must be careful with connections; sparking can occur when the negative terminals of two speaker outputs are shorted together. And with neither terminal grounded, the chance of shorting speaker terminals to ground is doubled. But these are trade-offs: the use of balanced, floating outputs can reduce distortion.

As for the Sony's tuner section, the less said the better. It is a pitifully small module, just as in so many other A/V receivers, and was unable to bring in anything but strong local signals.

Measurements

The performance of the Sony's amplifiers is significantly below the level of even low-cost analog A/V receivers. The measured S/N was not good: on gain-linearity tests, I could not resolve below -80 dB. White noise could be heard at the speakers, and at high-volume settings it was clearly audible from my center listening chair. The noise went down at lower volume settings, indicating that its source comes before the digital volume control. This is surprising, since we'd expect most of the noise to originate after the volume control, at the digital noise shaper.

The digital amp did other strange things, perhaps as a result of the potential problems with fully digital class D amplifiers, discussed in the sidebar on the system's technology. With a —60 dB, 1 kHz test signal applied to the input of the AVD-S50ES, I saw IM. spurs in the output spectrum, possibly stemming from the switching amp. When a full-scale (0 dB) 1 kHz signal was applied, the output included an unusual series of odd harmonics; the third harmonic was down only 54 dB, while higher odd harmonics (seventh, ninth, etc.) were 80 dB down—all the way out to the 13th harmonic, with the 19th harmonic still present at —90 dB (Fig. 1). Nothing like this would ever be seen in an analog amplifier! The most likely cause is saturation of the teeny-weeny inductors in the RFI filter (see the sidebar on circuitry).

The AVD-S50ES soft clips. Distortion comes out of the noise floor at 30 watts (-60 dB), and 1% THD (-40 dB) is reached at 120 watts. At 6 kHz the THD is at -48 dB (0.4%) for 100 watts output into 8Ω. The high level of high-order harmonics, coupled with the need to make our measurements through a 22 kHz bandpass filter to reject the noise that pours out of this unit, limits our ability to determine what the THD is for frequencies above 6 kHz.

The PowerCube dynamic power test (Fig. 2) did not go well, perhaps because of the digital switching power supply's limitations discussed in the technology sidebar. The PowerCube shows that the Sony puts out about 32

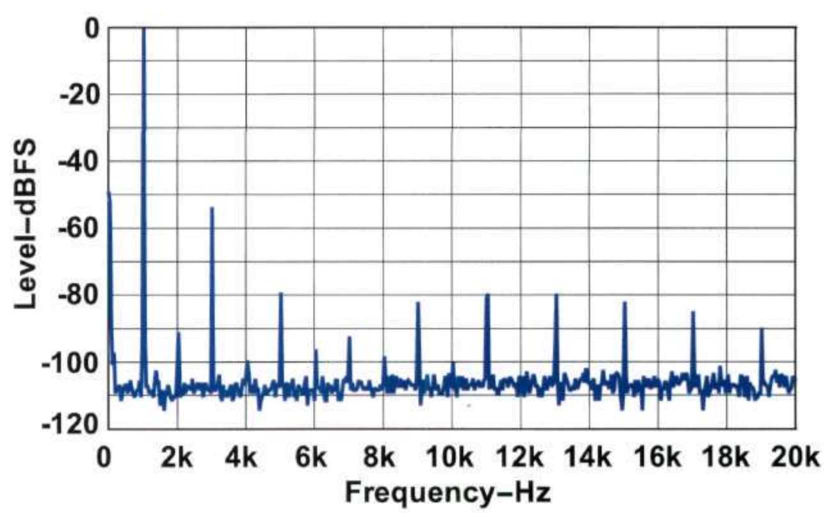


Fig. 1: Harmonics of a 1 kHz signal at 95 watts into an 8-ohm load.

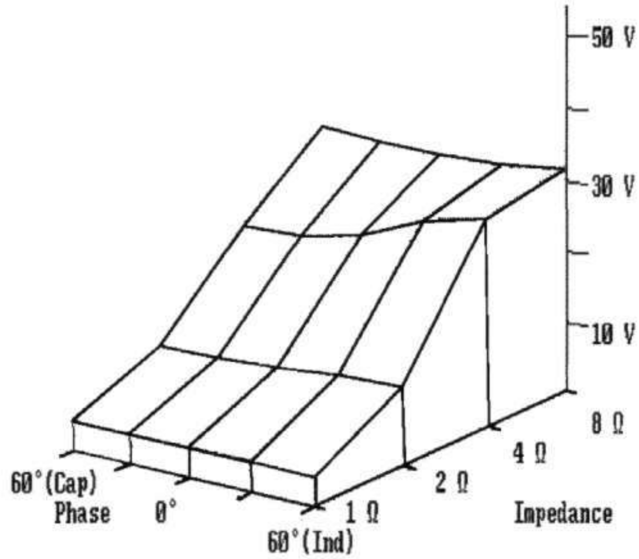


Fig. 2: PowerCube test of an AVD-S50ES amplifier channel. (See also PowerCube on page 34.)

V into all 8Ω loads. Capacitive 4Ω loads trigger the protection circuit to some extent; with loads sufficiently capacitive for a 60° phase angle, voltage dropped to 23 V. At 2Ω the voltage falls to 10 V, relatively independent of the phase angle. Into 1Ω, the Sony puts out 4 V.

The measured performance problems I found in this unit can be correlated directly with the chip design. The fact that the SACD signals are converted back into PCM before being turned into a different one-bit stream kills the only real advantage that a serious audiophile would hope to find in this unit. The lack of such important operating features as composite-video inputs and Dolby Pro Logic II Music adjustments is an additional reason not to recommend the AVD-S50ES to the advanced audiophile.

That said, the Sony would make most people very happy, because it is compact and eliminates the need for separate SACD and DVD player boxes. Sony has recently dropped the price of the AVD-S50ES from $800 to $500, at which it is an excellent value for the mass market. Its measured performance and feature list may not match those of two-box solutions at similar price points, but its audible performance should be about the same, and the convenience of having an SACD/DVD/CD/AV receiver that is fully interconnected in both the operational and electrical domains in a single small box is an overwhelmingly clear advantage. We caution those who seek surround-sound conversion of stereo sources as the primary reason for purchasing a new A/V system not to consider the unit, given its inability to allow adjustment of Dolby Pro Logic II Music. In addition, those who want to add in a DVD-A player in the future will not be able to do so with this unit.

The AVD-S50ES is the first receiver with an all-digital signal path and a built-in CD/SACD/DVD player. This is the future, and the AVD-S50ES represents the first commercial attempt to make the future happen. Unfortunately, though all its digital high technology makes the unit small, cuddly, and loveable, it cannot perform as well as the old analog designs.

—***David Rich***

BOX 978

(continued from page 4)

tube-versus-transistor sound. Son of a gun a second time! My tip-of-the-objective-hat to you and David and *Invention & Technology* magazine.

I would encourage your readers to obtain a copy of this particular issue of *Invention & Technology* and read the referenced article. The article is not without a minor fault (the author is way too easy on the tube cultists), but it does an excellent job of separating scientific fact from cultist bullshit.

As a subscriber to *The Audio Critic* and *Invention & Technology,* I can't help but mention that both magazines are published quarterly...!? Thanks for listening, Peter.

Regards,
Joseph M. Cierniak
Glen Burnie, MD

At the risk of sounding churlish in response to a flattering letter, let me ask you a couple of probing questions, foe. I know you are a regular contributor to The Sensible Sound, *as well as Editor/Publisher of the laser-printed mini-magazine* Sound Off, *but who appointed you to be the guardian and monitor of my "editorial integrity"? What did it matter to you? Did you actually spend time and energy tracking down Greg Keilty's telephone number? Don't you have anything more important to do? And would you have broken into a triumphal dance and written a self-righteous expose had you discovered that my story didn't jibe with Greg's? Gawd, what a busybody!*

Anyway, thanks for the tube-article reference and for your compliments.

—Ed.

Peripherals

ACCESSORY REVIEWS

By Ivan Berger, Guest Editor

Four Audio Side Dishes: Two Good Little Radios, the World's Best CD Rack, and a Switcher for Recordists

Two Good Little Radios

Boston Acoustics Recepter Radio & Tivoli Audio PAL

Boston Acoustics. 300 Jubilee Drive, Peabody, MA. 01960. Voice: (978) 538-5000. Fax: (978) 538-5199. E-mail: support@bostona.com. Web: www.bostona.com. Recepter Radio, $159.00. Tested sample on loan from manufacturer.

Tivoli Audio, 451 D Street, Suite 902, Boston, MA 02210. Voice: (877) 297-9479. Fax: (617) 464-0008. E-mail: mail@tivoliaudio.com. Web: www.tivoliaudio.com. PAL, $129.99. Tested sample on loan from manufacturer.

These two radios are descendants of Henry Kloss's KLH Model Eight, of 1960. The Eight had no frills, just a good tuner, a nice look and feel, a state-of-the-art small speaker from one of the world's top loudspeaker designers, and frequency contouring to wring the best possible sound from that speaker. The Tivoli Audio PAL, one of the last designs Kloss finished before his death last year, continues that heritage directly, right down to a vernier tuning dial like that on the Eight and all Kloss's later radio designs. The Boston Acoustics Recepter Radio is a new interpretation of the concept, from a company founded by people who had worked with Kloss. Each radio extends the concept in a new direction—portability in Tivoli's case, the practicality of a clock radio in Boston's.

As you'd expect from Kloss, the PAL (short for "Portable Audio Laboratory") is the one that breaks new ground. Most radios are horizontal designs, and all but the smallest portables have handles. The PAL sits upright and needs no handle because its case is cleverly designed for portability. Its vertical layout puts the Tivoli's narrow side up, for easier grasping, and its slip-resistant rubber coating has subtle grooves for a more secure grip. The radio's speaker enclosure is sealed to keep rain out and has captive flaps to weatherproof its rear-panel jacks when nothing is plugged into them. You shouldn't dunk this radio but it's okay to leave it in the rain—I've done it for weeks, without a problem. Leaving it too long in the sun, however, can affect the rubber coating (which comes in a choice of seven colors, from a dignified brown to a flashy yellow). The radio operates for about 15 hours per charge, depending on how loud you play it, and the green LED pilot light blinks when the NiMH battery's charge is getting low. Thoughtfully, Tivoli put its brand name on the plug-in charger, so it won't get confused with the wall-wart power supplies you already have.

Boston Audio's Recepter, the clock radio, has a more conventional layout and comes only in gray, with a narrower choice of faceplate colors: platinum, charcoal, or white. The digital clock and tuning dial glow a pale yellow green. With no need for weatherproofing, the speaker is vented through a port in the back.

Perhaps because the clock and display create heat even when the radio is off, the Boston radio has " feet, to help air circulate beneath it. However, because they're comparatively tall and narrow, these feet can get knocked off easily. The Tivoli's feet are shallow rubber pads that stayed put during the months I used it.

Portability aside, the Tivoli is the more versatile of the two. Although its built-in amplifier and speaker are monophonic, the radio's other circuitry is stereo, so you can listen binaurally through headphones or use the PAL as a tuner. (The Aux In and headphone jacks are in the back, together with a swiveling, telescoping antenna, and a jack for the battery charger.)

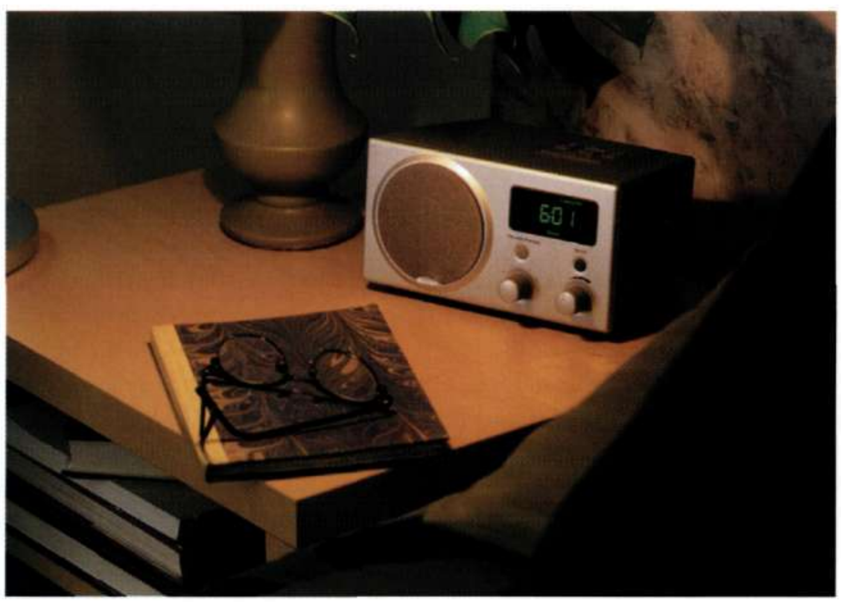
The only extra connections on the Boston Audio Recepter Radio are a 75-ohm FM antenna jack and spring clips for an external AM antenna, but it has several features the PAL does not. Its tuner has memory slots for up to 20 station presets, which can be whatever mix of AM and FM you like. Its two independent wake-up alarms can be set to rouse you with music, a buzzer, or both. (An AAA battery keeps the clock and station presets alive if the radio is unplugged or the power fails.)

The Tivoli's vernier tuning dial, Kloss's hallmark, consists of a geared-down dial-pointer ring surrounding a large knob. The knob's diameter (about 1½") makes fine adjustment easy; the dial pointer turns only one-fifth as fast as the knob does; and the gearing is damped to provide just the right amount of resistance for a pleasant tactile experience. The smooth vernier lets you tune slowly through a station to find its "sweet spot" on the dial instead of flicking past it; the AFC (automatic frequency control) has enough bite to enlarge that "sweet spot," yet is mild enough to let you tune weak stations even when a stronger one is nearby on the dial. Tuning this radio is *fun.*

Tuning the Recepter, though, is easier. Its digitally controlled tuner hits only the legal station frequencies, stepping 0.2 MHz per click in FM mode and 10 kHz per step in AM mode. The digital frequency display (which can be read from across the room) lets you dial in the station you want, even if you can't yet hear it, then fiddle with the antenna until that station comes in. And in Preset mode, of course, each click takes you to a station you've preselected. (An inconspicuous button above the tuning knob cycles through the three tuning modes.) Twenty presets are enough to store anyone's favorite stations (a whole family's, I suspect).

I mentioned earlier that the PAL's headphone jack enabled it to be used as a tuner. It would make a pretty good tuner, too. It's sensitive enough to pick up distant stations, selective enough to separate stations on adjacent frequen-

Tivoli Audio PAL



Boston Acoustics Recepter Radio

cies, and quiet. Reception improves, of course, with the antenna extended, and improves still more if it's reoriented for the specific station you want.

Using only the antennas supplied, the two radios' sensitivity and selectivity were just about exactly equal, with a slight advantage to the PAL because its adjustable antenna mast is easier to position than the Recepter's floppy wire. But the Recepter has connections for external antennas, with which it would probably outperform the PAL Interference rejection was excellent on both radios (even on AM) but the Boston's was better than the Tivoli's.

The Tivoli's sound is loud enough for outdoor use, clean enough for classical and other acoustic music, and punchy enough for reasonable rock. When the radio stands alone, it sounds a wee bit thin, but back it up against a wall or other surface (even the back of a laptop computer's screen will do) and the bass fills out pleasantly. The Boston radio has more volume and more bass, much of that likely due to its larger speaker (about 3¼" in diameter to the Tivoli's 2¼" or so). This is a small difference in diameter, but enough to give the Boston's speaker about twice the other's area.

The Tivoli Audio PAL is, like all of Kloss's radios, simple, sweet, practical, and charming. The charm comes mainly from its human engineering—the feel of its vernier tuning dial, and such niceties as flats on the small knobs to give your fingers a good purchase and let you see how they're set. And, like Kloss's other radios, it's an excellent performer.

The Boston Acoustic Recepter Radio is less involving, more businesslike. And it outperforms the PAL.

Yet, these two don't really compete with each other. The Tivoli won't wake you; the Boston can't join you outdoors, or even be carried from room to room without unplugging and replugging it, and it won't let you listen in stereo through headphones.

CD Storage

Davidson-Whitehall STORAdisc LS-576

Davidson-Whitehall, 290 M. L King Jr. Drive SE, Suite A5, Atlanta, GA 30312-21 00. Voice: (800) 848-9811 or (404) 658-1704. Fax: (404) 659-5041. E-mail: info@storadisc.com. Web: www.storadisc.com. Model LS-576, $625.00 to $1025.00, depending on finish. Tested sample purchased by author.

Davidson-Whitehall's STORAdisc is the best CD storage system in the world. Not the prettiest, hardly the most space-efficient, but by far the most practical. It's the only one I've seen that takes into account the way people actually access CDs.

The worst way to store CDs is in those storage racks that have one slot per disc. Once you have too many discs for easy retrieval from a random array, you need to organize your collection. Fill up a CD-slot rack with discs stored, say, alphabetically, and your growing collection becomes a problem: If you buy a new disc by, for example, Albeniz (or Audioslave), the only way to make room for it is to pull each disc in your collection out,

move it down one slot, and repeat with the next CD. Even if you leave a scattering of empty slots for expansion, you'll face this problem sooner or later.

Bookshelf-type storage is space-efficient and makes expanding your collection easy. When the top shelf fills, pull out a handful of discs and move them down one shelf, progressing shelf by shelf until you hit an empty space. You can leave empty spaces for expansion, but they can't be so wide that adjacent CDs fall over. And because the type on the jewel boxes' spines is small, you have to get down on the floor to see what's on the lower shelves.

The STORAdisc's clever design overcomes those limitations. The back of each shelf has a rubber stripe, providing enough friction so that even a single, isolated CD won't fall over. The shelves are angled up, making the spines of the jewel cases easier to read. And each shelf sits farther forward than the one above it, making it easy to get what you want even from the bottom shelf (whose front, on my tall STORAdisc, is more than 6 inches from the floor).

But being the world's best CD rack doesn't mean the STORAdisc is perfect. Because its lower shelves project further the lower they go, it's a lot deeper, front to back, than a straight CD bookshelf. The eight-shelf model LS-576, which holds (surprise!) 576 CDs, is 11 inches deep; it's also 31½ inches wide and 63½ inches high. The five-shelf LS-360 is 41¾ inches tall; single, stackable, 50- and 36-CD shelves are also available and are smaller in all three dimensions.

Some people may be bothered by the small but visible screw heads on the side panels. The reason the screws are visible is that you have to assemble the STORAdisc yourself, using a supplied Allen wrench. Though assembly took me a bit more than the half hour Davidson-Whitehall cites, it strained neither my back nor my brain. The finished rack is sturdier than any other furniture I've assembled myself.

Pricing depends on the STORAdisc's size and finish. The tall LS-576 costs $625.00 to $675.00 in textured paint, $850.00 in red or white oak, and $1025.00 in cherry, black walnut, pickled ash, and black or white lacquered oak. The LS-360 costs $495.00 to $820.00, depending on finish. Custom widths and finishes are available. The Davidson-Whitehall Web site gives unusually complete details.

Recorder Switchbox

Esoteric Sound Superconnector

Esoteric Sound, 4813 Wallbank Avenue, Downers Grove, IL 60515. Voice and fax: (630) 960-9137. E-mail: EsotericTT-@aol.com Web: www.esotericsound.com. Superconnector, $299.00. Tested sample on loan from manufacturer.

If you don't have a lot of recorders, you won't need this neat little switchbox, but I salivated at the thought of it. On my desk right now are a PC that doubles as a hard-disk recorder, a CD recorder, a DAT recorder, a cassette recorder, and an open-reel tape deck. Managing them all used to require plenty of plugging and unplugging, because the component I was recording onto one day often became be the source I was recording from the next.

The Superconnector eliminated all that connection-swapping—and the mistakes I sometimes used to make while doing it. This simple, passive switchbox, designed specifically for jungles of recording gear like mine, lets you dub from anything to anything else while you monitor any source or recorder that's hooked up to it. One-way switchboxes that add extra inputs to an audio system are common, but this is the only recorder-oriented switcher I know of that's currently in production.

The device is a small box, 10½ inches wide, 6 inches deep, and 1¾ inches (one rack space) high, mounted on a 19-inch rack panel. On the back are 17 pairs of gold-plated RCA jacks: inputs for three stereo sources (marked "Tuner," "CD," and "TV") plus input and output jacks for two processors (such as an equalizer and a noise reducer), four recorders (labeled "DAT," "Cassette," "Rcdrl," and "Rcdr2"), and your audio system's tape monitor loop ("Main Amp"). I have one more recorder than that, but no problem: on the front are three-conductor input and output phone jacks for a fifth recorder ("Ext").

Also on the front panel are two big knobs and three small toggle switches. The knob on the left selects which of nine sources you'll record from (the three rear-panel source inputs, the five recorders, or the feed from your audio system). The selected source is fed to all the recorders, so you can make up to five recordings of it at once. With the "Monitor" knob, on the right, you can select the output of any recorder or of the Superconnector itself.

Two of the toggle switches select processors (labeled "EQ1" and "EQ2"

on the front panel, though the corresponding jacks are labeled "Processor 1" and "Processor 2"). The third is a stereo/mono switch. Because all three switched circuits are in line with the record outputs, they affect the signal you're recording.

Frankly, I've never used the processor loops except to check that they worked properly. The stereo/mono switch, however, has been a godsend for dubbing from monophonic LPs and 78-rpm records. Stereo phono cartridges pick up the record groove's lateral and vertical undulations, but on monophonic records the vertical component is just noise. Switching to mono drops the noise level markedly. (This is no surprise, as most of Esoteric Sound's products are oriented toward record collectors.)

There is not much you can say about a passive component's performance. Either it screws up the sound, or it doesn't. The Superconnector doesn't. The rear panel has some gaps that theoretically could compromise the unit's shielding, but even with this switchbox sitting right next to my PC, I've heard no sign of interference.

The Superconnector has one severe potential problem (which the manual explicitly warns about): If you inadvertently set its "Source" switch to play a recorder whose own output selector is set to "Source," you'll create a feedback loop. The loud squeal this will send through your speakers could damage them—not to mention your relations with your neighbors.

Otherwise, the Superconnector is a delight. The controls work logically and are clearly marked, and the jack identifications are printed clearly on top of the chassis. I like the feel of the plastic-covered toggle switches but not of the square-ribbed knobs—admittedly a quibble. For my desktop setup, the projecting rack ears are a nuisance, nearly doubling the Superconnector's width. But for home studios, the obvious intended market, they're a necessity.

To some, the price may seem a bit high for a component that's nothing but a bunch of jacks and switches. However, because the market for a specialized switcher like this is small, the Superconnector has to be hand-built. And having built switchers for my own use, I know how much work is involved.

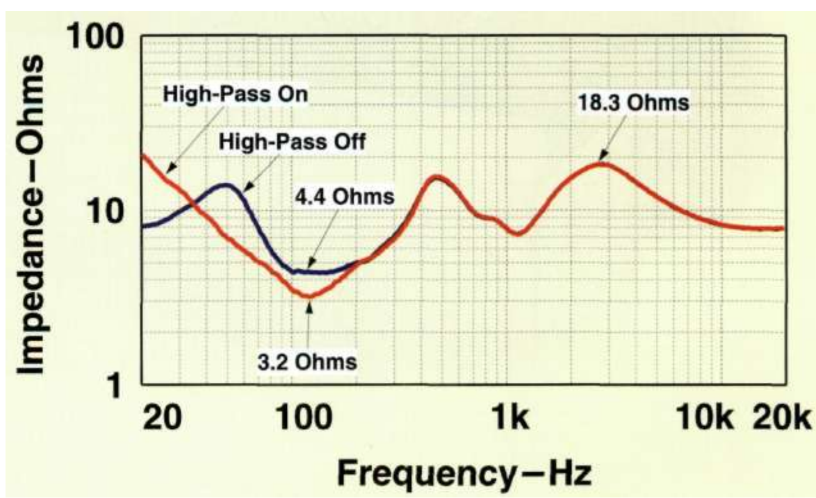
So: cheap at the price-if you need one.

—Ivan Berger

CORRECTION

In the Infinity Intermezzo 4.1t review (Issue No. 28, p. 16), the stated price of $3,500.00 per pair should have been $3,500.00 each-a lamentably easy error to make when pricing speakers, but lamentable nonetheless. Since then, the price has come down; in a mail-order catalog we recently saw "regular $5,000/pair, now only $3,995.00/pair."

In the same review, the graph captioned "Fig. 5: Impedance magnitude" was actually an inverted version of the impedance phase graph shown (correctly) as Fig. 6. The text, however, correctly described the impedance magnitude, and the correct graph is reproduced below.



Impedance magnitude of Infinity Intermezzo 4.1t (corrected graph)

"You want me to come up to your place to listen to your stereo? You don't have 5.1 surround sound with Dolby Pro Logic II and DTS 96/24? Get lost, retro boy!"

Peter and Tom Aczel

By Tom Nousaine

Audio's Top Urban Legend

In my last column, on audio's urban legends, I failed to mention perhaps the most persistent and seductive one: that new audio components—even cables—must be broken in, or "burned in," before they reach their full potential.

This gives the dealer an out when a customer brings a new component home and calls back to complain that it doesn't sound as good as he had hoped: "It'll sound better once it's burned in," the dealer will reply. And that advice may be useful for resolving unjustified cases of buyer's remorse—as long as the recommended break-in interval doesn't exceed the return period.

More to the point, do products really need breaking in? And if they do, how do they know exactly when to stop? Vinyl records have a wear cycle, but it doesn't stop; the discs simply wear out. Same with vacuum tubes, phono-cartridge suspensions, styli, CD or tape-recorder drive belts, and anything else having elastic or moving parts. These products break *down,* not *in.*

Solid-state electronic devices have no parts that break in. They either die very young (which the trade calls "infant mortality") or go on virtually unchanged for years. So if a solid-state amplifier turns on and makes sound the first time you turn it on, it is generally good to go. And cables, completely passive transmission devices, have nothing that can break in or needs to. Audio buffs often talk about the benefits of break-in, and they have a reason (which I'll get to later) — but not the reason they think.

What about speakers? Unlike amps and cables, speakers do have elastic elements and moving parts. If anything should need breaking in, speakers should. And manufacturers often do recommend substantial break-in intervals. One maker, for example, suggests "at least 50 hours at moderately loud levels. . . even more improvement after 100 hours of playing." I asked a local retailer about his store's return policy. He explained that all his customers had an opportunity to listen to fully broken-in speakers on the floor and sometimes with weekend loans, but once they purchase they have 15 days to return it (and must pay a 15% restocking charge). By my reckoning, a buyer of the speaker I just cited would have to listen at a loud level for 3½ hours every day for two weeks just to break his speakers in, and would need another two weeks to get that "even more improvement." Of course, he could also just leave them on all day for a few days—as long as there's nobody home to be bothered.

Let's think about this in engineering terms: It would appear that the drivers most likely to need breaking in would be woofers, which have compliant suspensions and relatively floppy cones. Over the years, I've conducted two in-depth studies of 12-inch woofer break-in, both at the behest of manufacturers who insisted their products *had* to be broken in for lengthy periods before being reviewed. The first time, I measured a driver before and immediately after a long break-in period, and found that its free-air resonance frequency (f_S) had fallen by five to ten percent and compliance, or springiness (C_{AS}) had increased by a corresponding amount. The driver's calculated Q values changed accordingly.

However, using a computer program to model an ideal enclosure for this driver gave me the same results whether I entered the data for the driver I'd "broken in" or for a fresh one. In other words, the lowered free-air resonance counteracted the increased compliance to give me the same results. I built a pair of the enclosures the computer had recommended, and installed a fresh driver in one and the broken-in driver in the other, then made the same performance measurements for each. My results for the two speakers were not quite identical, but the differences were within the tolerances implied by the unit-to-unit variation of four samples I had measured fresh. The two speakers sounded the same, too.

I repeated this experiment for another manufacturer, who insisted his speaker required at least a 48-hour break-in. In this case I broke the driver in while it was in the manufacturer's recommended enclosure. The speaker's frequency response was the same before and after break-in. This time I also measured the resonant frequency (f_{SB}) of the system at intervals (30 minutes, 1, 2, 3 hours, and the next day) following the break-in cycle. The f_{SB} did change but, interestingly, it slowly dropped back to its original value once the speaker had cooled down for a few hours. The woofer had warmed up but its performance hadn't changed—it hadn't really broken in. If you truly want to get whatever effects (if any) might result from breaking a speaker in, you'll have to warm it up for a couple of days before listening sessions or make sure it never cools down.

Despite all this, there *is* a break-in period for drivers. But according to a transducer engineer who used to work for a large American maker of drivers and finished speaker systems, the whole deal takes just a few seconds, and usually occurs during the final quality-control check at the end of the driver assembly line.

So do speakers break in once we get them home? Hell, no. And we should be thankful for this. I'd worry about a manufacturer who'd let a product leave his factory before he'd verified its final performance. If breaking in is truly needed, it should be done at the factory.

If a product—speaker or otherwise—sounds substandard, it won't improve with breaking in. But you might think it has, because *you* get broken in, acclimating to to the product's sound over time. Humans are remarkably adept at adapting to any stimulus. For example, a fan may sound loud when you turn it on, but you may not even notice the noise after you've been hearing it for half an hour. Turn it off, though, and the room will sound remarkably quiet—until you get used to *that* noise level. While you shouldn't expect audio products to improve with use, you shouldn't forget that what changes may be *you,* not the product.

Finally, if you insist on breaking-in your speakers, do it carefully. Letting the speaker play overnight with a low-level test signal or music is prudent. Do not play a noise, sine wave, or other continuous signal through your speakers at high levels for an extended time. That's for the speakers' sake as well as for your ears'. There is a technique to minimize the aural annoyance during break-in, by placing the speakers face to face but wiring one with reversed polarity, so that the sounds from the two speakers largely cancel. But this technique, which encourages driving the speakers at high levels for a long time, is fraught with danger: too much level for too long, and you'll become intimate with the smell of melting voice-coil glue. The only thing that has in common with the smell of good perfume is the expense.

Next issue: True or false? Every room is different. Rooms may all be different, but they are more alike in important ways than is commonly believed.

Bose Quiet-Comfort 2 Headphones

(continued from page 32)

the listener's head. (Those cushions, and a well-designed headband, also make the headset comfortable to wear for long listening sessions.)

I've flown with the earlier model and can attest that the noise control works fine, and I do indeed get off the plane feeling less fatigued than I do after a couple of hours with my ears naked to the noise. I'd expect the QuietComfort 2 to handle noise equally well. In fact, during the press demo, the presenter sneakily fed a recording of airliner noise through speakers at 83.5 dB SPL while we were listening to music through the phones; none of us noticed until we turned the headphones off. Bose even provided for people who want to use the headphones just to counter ambient noise, without listening to music, by making the cord detachable; in the original model, with the electronics dangling from the cord, that wasn't possible.

The QuietComfort 2 doesn't sound as live and airy as the best open-air phones, but the extra silence it surrounds the music with is a good trade. The highs were clear and extended, the lows were rich and deep, and both stayed in balance with the midrange—nothing quite exceptional but nothing to take exception with. The sound was very neutral—more appealing, I suspect, to listeners who preferred the old Shure V-15 phono cartridge than to those who preferred its moving-coil competitors.

In designing the QuietComfort 2, Bose made a series of tests with tiny in-ear microphones "to identify how your usual perception of sound changes when you put on a pair of headphones," and used this data in the development of "proprietary acoustic equalization techniques" to make the sound a closer match to that heard from speakers. (No attempt was made to compensate for the perspective change between headphones and speakers—centered soloists are still inside your forehead.) To me, the new model does sound just a bit more natural than the original QuietComfort, but some listeners may prefer the older model's more emphatic bass.

The QuietComfort 2 is a definite improvement on a product that was already good. It offers the same comfort and isolation from noise as the original QuietComfort, marginally better sound, and far greater convenience. These differences are well worth the additional $50 they cost.

An open mind is all very well in its way, hut it ought not to he so open that there is no keeping anything in or out of it. It should be capable of shutting its doors sometimes, or it may he found a little drafty.

—SAMUEL BUTLER (1835-1902)

Capsule CD Reviews
(including SACD and DVD-V)
By Peter Aczel, Editor

Since I am in the process of curtailing my contributions to this journal, with 90% retirement as my wistful goal, I would welcome another reviewer to take over this column, if I could only find the right one. He or she would have to be not only reasonably knowledgeable and highly enthusiastic about music but also genuinely finicky about audio quality—and that's the problem. Most music critics don't distinguish between pretty good, better, and superb recorded sound—usually they don't even own first-class audio equipment—and this is an audio publication. Eventually I'll find a solution, especially since the recording technology has reached the point where CDs are almost uniformly good and thus require less and less technical criticism. (David Ranada filled the bill years ago, but he is much too busy these days.) As for the reviews below, please note that the year in parentheses after the CD number is the year of recording, not the year of release.

BBC Opus Arte

This has turned out to be a reliable label for first-rate English opera productions. Distributed by Naxos.

Giuseppe Verdi: *II Trovatore. José Cura, Manrico; Dmitri Hvorostovsky, Count di Luna; Verónica Villarroel, Leonora; Yvonne Naef, Azucena. The Orchestra and Chorus of the Royal Opera House (Covent Garden), Carlo Rizzi, conductor. DVD-Video OA 0849 D (2002).*

Caruso once quipped that it is very simple to cast *Il Trovatore*—all you need is the four greatest singers in the world. This production doesn't quite satisfy that requirement, but it's very good. As a matter of fact, there may not be a better baritone in the world today to sing Count di Luna than Dmitri Hvorostovsky, and the other principals are also highly competent or better. The swarthy Argentinean José Cura even looks like a Gypsy soldier/musician. He is a fine tenor who strains occasionally on the high notes but is very convincing overall (although he sings only one verse of *Di quellapira*—chicken!). I have nothing but praise for Yvonne Naef as Azucena, a beautiful performance, and Verónica Villarroel, looking a little older than Leonora should, is also excellent once she has warmed up her voice. It's Hvorostovsky, however, who is truly world-class, a mesmerizing singer who totally dominates his scenes. Rizzi's conducting is thoroughly idiomatic, and the orchestra is above reproach, never submerged under the voices, although you see very little of the instrumentalists with the strangely restricted camera angles used. The staging is somewhat peculiar; the costumes and props are closer to Napoleonic than 15 th century Spanish, and the background scenery is all over the place, sometimes period, sometimes industrial. The picture is 16:9 anamorphic; the 5.1 Dolby Digital surround sound is fine but does not by any means set a new standard. All in all, I would grade this effort between B+ and A-.

The Cleveland Orchestra

This is the orchestra's private label, releasing recordings of its own performances only.

"Christoph von Dohnányi: Compact Disc Edition." *Archival live stereo recordings of the Cleveland Orchestra's home concerts, 1984—2001, comprising 28 works by 26 composers, conducted by Christoph von Dohnányi. MAA-01032-A/B/C (10 CDs, previously unreleased).*

Christoph von Dohnányi began his association with the Cleveland Orchestra in 1982 and ended it in 2002, having been the orchestra's music director for the last 18 of those 20 years. He basically owns that stupendous orchestra, one of the finest in the world. This monumental set of CDs, taped by the orchestra in live analog stereo for radio station WCLV, appears to be quite competitive sonically with the Cleveland's commercial digital releases and is a fit celebration of two decades of glorious music-making. Half of the recorded works were composed in the 20th century; the rest are 19th-century classics from Beethoven to early Mahler, with the exception of one late-18 th century symphony by Haydn, but no Bach, no Mozart. The performances range from routinely first-rate to superb (e.g., an absolutely stunning rendition of that old warhorse, Liszt's *Les Préludes).* The picture of the maestro that emerges from the recordings is that of a musician of exceptionally wide-ranging tastes who is in total, hair-trigger control of his magnificent band. Few persons will want to pay $175.00 (plus $12.00 S&H) for the set, which includes a 72-page booklet, but maybe the Cleveland Orchestra will eventually release the ten CDs one by one. For reviewers who received a complimentary copy, the set is certainly a blast.

Delos

John Eargle, one of the best recording engineers in the world, has retired; his successor at Delos appears to be his assistant, Jeff Mee, but lately most Delos recordings come from Russia, recorded by Russian engineers.

Ernest Bloch: *Prayer, From Jewish Life, No. 1; Schelomo.* ***Nina Kotova:*** *Concerto for Cello and Orchestra (recording premiere).* ***Max Bruch:*** *Kol Nidrei (Adagio for Cello and Orchestra), Op. 47. Nina Kotova, cello; Philharmonia of Russia, Constantine Orbelian, conductor. DE 3305 (2001).*

This I haven't seen before. I didn't even know it existed. Imagine a top-notch virtuoso cellist and composer who is a real babe, a gorgeous ex-model! Only in America.. .well, actually she is a native Russian, born in 1971 and transplanted to America only about 12 years ago. She plays the cello with an exceptionally rich, burnished tone and an utterly secure technique. Her own composition is accessible-contemporary in idiom, exploring every possible nuance of cello sonority, and somewhat pedestrian in orchestration. (You expected this dishy "10" to orchestrate like Respighi, on top of everything else?) It's too soon to form a strong opinion about the concerto, one way or the other. The Bloch and Bruch pieces are familiar and not really my cup of tea, but they are beautifully (and idiomatically) played. The audio is excellent, no problem, but the music doesn't really test the mettle of the Russian recording team, so I can't tell whether we need John Eargle back. In any event, Nina Kotova is headed for superstar status.

EMI Classics

This British conglomerate appears to be still able to make frequent recordings of great orchestras under important conductors, something that has become unaffordable in America.

Gustav Mahler: *Symphony No. 5 in C-sharp Minor. Berlin Philharmonic, Sir Simon Rattle, conductor. 7243 5 57385 2 3 (2002).*

I have reviewed one or more Mahler symphonies in six of the last seven issues, not because Mahler is my favorite composer (I still have certain reservations about the breast-beating style and prolixity of some of his works) but because, on account of his orchestration, he is *the* composer for the hi-fi era—and this is an audio magazine. This new version of the Fifth is probably the best digital (meaning: since the mid-'80s) recording of them all. Rattle and the great orchestra of which he recently became the leader give a more nuanced, coherent performance of the symphony than any other I can recall. Those others seem episodic by comparison, a sequence of sound effects, whereas Rattle shapes the work like a Haydn or Mozart symphony, with every note related to the previous one and everything in perfect balance. Very impressive. On top of it, even though EMI is certainly not an audiophile label, the recorded sound (by a British, not German, team) is of demo quality—it couldn't be better. Performed and recorded this way, the Mahler Fifth really sings and hangs together.

W. A. Mozart: *Violin Concerto No. 3 in G Major, K. 216; Adagio and Fugue in C Minor, K. 546; Symphony No. 41 in C Major ("Jupiter"), K. 551. Itzhak Perlman, violin; Berlin Philharmonic, Itzhak Perlman, conductor. 7243 5 57418 2 0 (2002).*

Anything recorded by Itzhak Perlman is worth listening to, and this CD is no exception. The concerto is played with the superstar's wonted magnificent tone and musicality, but there exists an early digital recording young Perlman made with James Levine and the Vienna Philarmonic (Deutsche Grammophon, 1983) that is much sprightlier, more variegated, and generally more exciting. The present performance sounds somehow homogenized by comparison. It helps to have a good conductor when you're busy playing the fiddle, and youth is sometimes an advantage. As for the symphony, the competition is enormous, and this is just another decent performance without any truly memorable qualities. Perlman is obviously a fine musician but not a major conductor. The audio quality is routinely good but not in the same league with the Mahler above, recorded in the same hall.

Harmonia Mundi

I keep returning to this label for solid, old-fashioned quality in a classical-music world where marketing rules and quality is random, if not beside the point.

J. S. Bach: *The Art of the Fugue, BWV1080. Fretwork: Richard Campbell, Richard Boothby, Wendy Gillespie, William Hunt, Julia Hodgson, Susanna Pell, viols. HMU 907296(2001).*

Fact: Bach didn't call his final incomplete masterpiece *Die Kunst der Fuga;* his pupil and son-in-law J. C. Altnikol did. Bach just wrote *Contrapunctus* on the title page. Fact: There is no indication of instrumentation in the open score, and far from all experts agree that it was intended to be a keyboard work. Opinion: Six viols producing a beautifully blended sound are a better instrumental combination than most to play this music. This is a gorgeous-sounding performance of difficult, highly abstract music, best listened to one fugue at a time. Some find the unvarying D Minor of the 20 fugues monotonous; others are convinced that this is the most sublime music in the world. Both opinions have validity. Certainly *Die Kunst* represents the absolute pinnacle of contrapuntal composition. Personally I'd rather listen to a late Beethoven quartet when I'm in the mood for serious fare, but you may very well disagree. The six English musicians who constitute Fretwork are world-class string players, and the recording does them full justice. Highly recommended to a necessarily limited circle of music lovers.

Bela Bartók: *The Miraculous Mandarin, Op. 19, Sz. 73; Dance Suite, Sz. 77; Four Pieces for Orchestra, Op. 12, Sz. 51. Orchestre National de Lyon, David Robertson, conductor. HMC 901777 (2001).*

The *Mandarin,* Bartók's 1918-24 masterpiece (you could argue it's his *Sacre,* with its savage rhythms and dissonances), has not been performed quite so authoritatively for many decades. In 1999, Peter Bartók, the composer's younger son, restored the work's original version, which had been hacked, cut, and edited to pieces over the years. (The only change from the original is a modified ending that Bartók himself composed between 1926 and 1931.) This is the premiere recording of the restored version, complete with corrected dynamics, bowings, etc. The American conductor David Robertson leads a brilliant performance, illuminating the work's remarkable coloristic details, not just its brutality, and the Lyon orchestra plays magnificently, proving itself a world-class ensemble. The other two Bartók compositions are of no lesser importance (Bartók wrote no mediocre music) and are equally well played. The audio quality of the disc is on the same high level, with an extremely wide dynamic range and low distortion. This CD is a must for Bartók lovers.

Arias for Farinelli. *Vivica Genaux, mezzo-soprano; Akademie für Alte Musik Berlin, René Jacobs, conductor. HMC 901778 (2002).*

Carlo Broschi detto Farinelli (1705-1782) was a castrato and the most adulated, almost deified, singer of his time. He sang with an astonishingly perfect vocal technique in an extremely virtuosic, highly ornamented style that was already passé in Mozart's days. Forgotten composers, such as Porpora, Giacomelli, Hasse, revived for this recording, were some of his repertory. The Alaskan-born mezzo-soprano Vivica Genaux is one of the few practitioners of this forgotten art—and she is amazing. You could argue that the almost ridiculously florid music is a bit monotonous—when you've heard one aria you've more or less heard them all—but it is still very exciting when sung with this kind of virtuosity. Rene Jacobs was the conductor of the *Così fan tutte* I raved about a number of issues ago, and here once again he makes period practice vital and relevant.

Such attacks, such crispness . . . The recorded sound is extremely clean and sparkling, as good as it gets. Not a CD for everybody but far from negligible.

Mapleshade

Pierre Sprey continues to baffle me with his cockamamie two-track analog taping technique, with which he achieves the most astounding you-are-there realism in playback. Hey, the outcome is what counts, not the means.

"Gentle Giant of the Tenor Sax." *Bob Kindred, tenor sax; Larry Willis, piano. 09032 (2001).*

All the giants of the tenor sax from the heyday of modern jazz are dead. The aging Bob Kindred continues the tradition and occasionally rises to their level on this CD. Is he a John Coltrane or a Stan Getz? I don't think so, but Pierre Sprey, the head of Mapleshade, does, and some reputable critics concur. At any rate, he is awfully good, the genuine article. You don't hear playing like this nowadays, so I could be wrong. The recorded sound is unbelievable and alone worth the price of admission.

Naxos

No longer just the leading low-priced classical label, Naxos is now the leading classical label, period, based on the number of new releases every month. It's amazing how successful their marketing plan of selling lesser-known artists with low performance fees but high performance has been. The big names have meanwhile priced themselves out of the hurting classical market. Of course, if you want big names—old big names—there is Naxos Historical.

J.S. Bach: *Cello Suites (Complete). Alexander Rudin, cello. 8.555992-93 (2 CDs, 2000).*

Playing polyphonic music on a single unaccompanied stringed instrument is closer to a circus act than to musical practice; yet just about every cellist of some standing has recorded these six suites of baroque dances, if for no other reason than as a test of technique. I haven't listened to them all, not even every fourth one, but I can testify that the brilliant Alexander Rudin passes the test with flying colors. He also manages to sound warmly musical in the process. More than that I dare not say because no one can really *soar* in this music. It is just too damned difficult and awkward. It's enough that he sounds.. .well.. .at ease, comfortable. His technique rises above the hardest passages. Some find the Bach cello suites to be utterly sublime music, but let's face it, they cannot possibly give you the same thrill as, say, the organ Toccata and Fugue in D Minor—the instrumental format militates against it. This recording does provide audio thrills, however; you can close your eyes and pretend the cello is there, in the room with you. The sound is that good.

Feodor Chaliapin: A Vocal Portrait. *Naxos Historical, Great Singers series. 8.110748-49 (2 CDs, 1907-36).*

The Russian basso Feodor Ivanovich Chaliapin (1873-1938) was arguably the greatest operatic bass of all time, certainly of the phonograph era. His voice was huge, effortless, absolutely even in scale, and virtually without vibrato—a unique instrument. His histrionic ability was also of the highest order, sometimes overpowering the composer's intentions; he certainly didn't sing everything exactly as written but was unfailingly fascinating. Here one CD is devoted to his acoustic recordings, up to 1924, the other to his electrical recordings, ending in 1936, when he was 63 and almost as good as ever. Altogether there are 40 tracks on the two CDs, superbly remastered by the blind specialist Ward Marston, so that even the oldest selection (1902) is entirely listenable. As a matter of fact, Chaliapin's deep bass mated better with the acoustic recording technique than, for example, Caruso's higher voice, so that the old tracks have been easier to restore. The music in this collection is mainly 19th century opera and song, with a heavy sprinkling of Russian composers, Mussorgsky in particular. You don't know the possibilities of the bass voice until you have heard these two CDs.

Franz Schubert: *Sonata No. 20 in A Major, D. 959; Sonata No. 15 in C Major ("Reliquie"), D. 840. Jenö Jandó, piano. 8.554470 (1998).*

Jandó is Naxos's house pianist, one of those affordable first-rate artists referred to above, and at least in the classics (namely Mozart, Beethoven, Schubert, etc.) he is as good as anyone with a bigger name—and I mean anyone. In the D. 959 sonata, one of the three from Schubert's miraculous last summer (1828), he plays spellbindingly, making the 40 minutes of heavenly music appear too short because you want it go on forever. Both technically and musically, he performs on the highest level. To me this is one of those desert-island-top-20 pieces, too beautiful for words. The unfinished "Reliquie" sonata is merely a curiosity by comparison, not that it isn't very fine music. Jandó does it full justice. The recorded sound (Phoenix Studio, Budapest, no longer active) is absolutely state-of-the-art, none better. A great CD.

Richard Wagner: *Die Meistersinger von Nürnberg, Prelude to Act I; Götterdämmerung, Siegfried's Funeral Music (Act III); Tristan und Isolde, Prelude to Act I; Der fliegende Holländer, Overture; Tannhäuser, Overture (Dresden version); Lohengrin, Prelude to Act III; Siegfried Idyll. Berlin State Opera Orchestra, Karl Muck, conductor. Naxos Historical, Great Conductors series. 8.110858(1927-29).*

Karl Muck (1859-1940) was 8 years older than Toscanini and regarded in his time as one of the two or three greatest living Wagner conductors. He was 24 years old when Wagner died, so he was completely aware of contemporary performance styles. In these marvelous restorations by Mark Obert-Thorn of early electrical recordings, Muck plays Wagner as if the composer were still alive, without any fuss, bother, mannerisms, or religious awe, just total focus and accuracy. The results are absolutely thrilling. The *Meistersinger* overture, for example, is played much faster than is today's reverent/pompous practice, and it works. The timpani strokes announcing Siegfried's death are much louder than in today's recordings, perhaps because of the limited dynamic range, and it also works. The *Tannhäuser* overture is again faster than usual, and I think I like it better that way. There is a lack of ceremony about these performances that is most refreshing. They are simple and noble instead of worshipfully "interpreted." The audio quality is amazingly lifelike for 1927-29; I can't recall any recordings of comparable fidelity until well into the 1930s. Another triumph for Naxos Historical.

RCA Victor Red Seal

After 100 years, still the label of some of the world's greatest musicians, although its most glorious era is a thing of the past.

Robert Schumann: *Sonata No. 1 in F-sharp Minor, Op. 11; Carnaval, Op. 9. Evgeny Kissin, piano. 09026-63885-2 (2001).*

At 30, when this recording was made, and now 31, Kissin is without question the world's No. 1 *young* pianist and perhaps even the world's No. 1 pianist, period, at least of the Rachmaninoff/Horowitz (as against the Schnabel) kind. His control of the keyboard is so absolute and so finely tuned that I don't understand critics who don't see the distance he has put between himself and the rest of the field. Not that this gives him the interpretive advantage over everyone else in these Schumann pieces; they can be played differently and just as impressively; but his clarification of the sonata's textures is without equal, as is his virtuosity in the many various very short episodes of *Carnaval.* He may not be the most *musical of all* pianists but he is the most explicit. I predict that when he mellows (maybe at 40?), he will have it all and rule the world. These works of Schumann, by the way, are among his very best, more interesting in my opinion than any of his later compositions, when he was slowly approaching insanity. The recorded sound of the piano, played in a German studio, is resonant and incisive, with close to optimum balance between warmth and clarity.

San Francisco Symphony

This orchestra has started to issue its recordings under its own label, just like the Cleveland Orchestra. Economy, no doubt.

Gustav Mahler: *Symphony No. 1 in D Major ("Titan"). San Francisco Symphony, Michael Tilson Thomas, conductor. 821936-0002-2 (SACD, 2001).*

At the risk of overloading this column with Mahler symphonies, I had to include this one because it represents a major exception to my frequently mentioned reservations about Mahler. The First is a highly original, fresh, sparkling, unmannered work of young genius, no question about it. I love it without qualifications. On top of it, Thomas's interpretation is the best one 1 can remember since the beginning of the digital era. It is straight, natural, meticulous but unfussy, just beautifully musical. Gorgeous playing. The SACD surround sound is excellent but for some reason 5.0 instead of 5.1; the subwoofer channel has no feed. I consider that a drawback since some surround-sound systems (not mine) consist of five small speakers plus subwoofer. In that case—no bass.

Sony Classical

I am still in the habit of bracketing this label with its predecessor, CBS Masterworks, but I guess the identity is fading.

Salvatore Licitra: The Debut. *Arias from* ***Puccini*** *and* ***Verdi*** *operas. Salvatore Licitra, tenor; London Symphony Orchestra, Carlo Rizzi, conductor. SK 89923 (2002).*

Here he is, Pavarotti's alleged successor—and the claim is not without solid foundation. Young Licitra is a stupendous tenor, with a huge, effortless voice, unstrained top notes, and a musician's ear, ready to wow the world. Is he as good as young (as distinct from aging) Pavarotti? Perhaps not quite as polished; in my opinion the Pavarotti of the late '60s and early '70s was second only to Caruso in the 20th century—better than Gigli, better than Bjoerling, better than Corelli, better than anyone else—and only later went into an up-and-down phase (mostly down). Licitra comes close, though; maybe he just needs a few more years to be a legend (he is 34 years old). He could perhaps further improve the evenness of his scale and cut down on the sobs, but overall his singing is better than that of 99% of all tenors. The program here includes all the warhorses—*E lucevan le stelle, Nessun dorma, Celeste Aida,* etc.—inviting comparison with the greatest of the great. He passes. His high notes, in particular, are of awesome power and absolutely clean. Carlo Rizzi and the London orchestra provide excellent accompaniment, and the recorded sound is all it should be.

Telarc

This label has become the major source for multichannel SACDs of classical music and jazz—and therefore, since DVD-Audio is stagnating, the major source for classical/jazz multichannel recordings in general. That the latter are superior in sound quality to stereo recordings is now no longer debatable; it is an obvious, accepted fact.

Reinhold Glière: *Symphony No. 3, Op. 42 ("Il'ya Murometz"). London Symphony Orchestra, Leon Botstein, conductor. SACD-60609 (2002).*

I don't think this is much above the level of pretty good movie music—great orchestration, thin substance—but the multichannel recording is state-of-the-art, in transparency, envelopment, localization, the whole bit. The venue, Watford Town Hall in England, may have had something to do with that. The London Symphony Orchestra plays beautifully, as usual; Botstein's conducting is routine. One for your sound-demo collection.

Gustav Mahler: *Symphony No. 6 in A Minor ("Tragic"). Philharmonia Orchestra, Benjamin Zander, conductor. Also: Benjamin Zander Discusses Mahler's Sixth Symphony. 3 SACD-60586 (3 SACDs, 2001).*

Some critics consider this Mahler's best symphony. I'm not one of them; I find too many of the ranting, overheated, and to my mind insincere elements in the music that make me hesitate to rank Mahler among the greatest of the great. That it is an interesting, complex, and far from negligible work is of course undeniable. Zander is true to form in his interpretation, rendering a superbly detailed and accurate reading of the score, close to what I imagine Mahler had in mind. Jack Renner's multichannel recording is stupendous, an exploration of all the possibilities of the medium, and alone worth the price of admission. It will make you conclude that stereo is dead. As for Zander's exegesis of the symphony, we know from previous occasions that he is uniquely qualified and fascinating. A monumental set of discs, no matter where on the Mahler scale you rate it.

Jean Sibelius: *Symphony No. 2 in D Major, Op. 43.* ***Eduard Tubin:*** *Symphony No. 5 in B Minor. Cincinnati Symphony Orchestra, Paavo Järvi, conductor. SACD-60585 (2001).*

I don't have an opinion—requiring several hearings—on the 1946 Tubin symphony, but I can report that the Sibelius is very well played, in a crisp rather than grandly romantic performance. The socko finale is rousing enough, at any rate. Paavo Järvi (Neeme Järvi's son) is new to the Cincinnati, and it appears to be a happy partnership. Michael Bishop's multichannel recording is outstanding; if you have a 5.1 system, you won't be tempted to switch to stereo, I promise you.

TAC