CAN A CAR RADIO OUTPERFORM A HIGH-END FM TUNER?

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ISSUE NO. 26
Display until arrival
of Issue No. 27.

The Audio Critic

The 10 Biggest Lies in Audio

Also in this issue:

Loudspeakers, including a $229 system that sounds like $15,000 (but don't buy it for your listening room).

Power amplifiers, D/A converters, AV electronics, a TV, and more.

Plus our regular features, columns, letters to the Editor, CD and DVD reviews—and our new look!

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ISSUE NO.26 • FALL 2000

What Did I Tell You?

Issue after inevitably delayed issue, I have been talking about our need to form a partnership with a publisher. Not just any publisher but one that could help us publish regularly and profitably without interfering with our editorial policy. Well, I have found such a publisher, and you are holding the earliest fruit of our partnership in your hand. Looks a lot more major-league, doesn't it? More importantly, it will be followed by the next issue, and the one after that, and all the others after that, at quarterly intervals, as originally intended. We are even talking about a bimonthly schedule in a year or two.

Our new partner is The CM Group, based in Toronto (haven't I always professed Canadophilia?) and headed by Greg Keilty, a widely recognized circulation expert. I have retained complete editorial autonomy. This publication will remain a consumer advocate and not one of the compliant handmaidens of the audio industry. You will notice that in this issue we are, to some extent, still playing catch-up because we had to clear our pipeline of accumulated products that had been submitted for review. That will no longer be the case in our next issue.

My only worry is that some of our readers have become accustomed to our former double issues for the price of one—overstuffed out of guilt by your forever tardy Editor—and will now want the same bargain every 90 days. Sorry, guys, that may have been a bonanza for *you*, but it was a hell of a way to run a magazine. Greg won't stand for it.

Peter Aczel

The Audio Critic

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BOX 978

Our fresh start with our new publisher, after the long interruption of our publishing schedule, leaves us with very few letters that are still of current relevance. The two below, from two of the most distinguished names in audio, remain required reading. Write us letters like that and there will be more pages again in this column. Address all editorial correspondence to the Editor, The Audio Critic, P.O. Box 978, Quakertown, PA 18951-0978.

The Audio Critic:

Thank you kindly for sending the most recent issue of *The Audio Critic.* I always enjoy reading your nice little magazine. It is a pleasure to see that you have managed after all these years to continue to speak the truth and keep your sense of proportion and humor.

I noticed your quest for a solution to "One Last Mystery." Though I trust there are more mysteries to come, I have wondered about the same matter over the years and thought I would make a few comments about it.

I have many time said that there are so many more fakes and frauds in the audio area than in many others, each of which should raise the same amount of passion. I am not familiar with the automotive field but I am very familiar with photography. There are two aspects to the relative sanity in the area of photography as compared to audio. One of the issues has to do with the ability of persons to make equipment and the other with the ability of persons to compare the performance of the equipment.

On the first issue, I would suggest that it is all but impossible for the photographic equipment field to be flooded with equipment designed by incompetents, fakes, and frauds, as is the case for audio equipment, because it is so difficult to do so. Anyone, qualified or not, can purchase the components necessary to make some sort of amplifier or preamplifier that will work, more or less. Anyone can put this stuff together with even minimal competence. There are magazines that describe construction and supply houses that provide the parts and instructions to make loudspeakers and electronics and so forth. There is in electronics and audio a long tradition, starting with the amateur radio groups, to build stuff. As a result, stuff is indeed built, and some go so far as to polish and glitz up their stuff, market it with technical drivel, and take in the suckers.

This is not so in the field of photography. There are no cameras being built by amateurs. There are no camera parts supply houses. There are no lens grinding kits and shutter parts vendors. There is no tradition of building a camera from parts. So, as we might expect, there are no purveyors of odd or silly cameras. Every photographer knows better than to be taken in by an advertisement for some sort of special super camera that would have magical properties. They know that such things are indeed silly.

As a result, photography is relatively free of fakes and frauds who push equipment with special properties, compared to the audio field. Photographers work at taking pictures, just as audio professionals work at making recordings.

There is another issue that I think is just as important, possibly more important. Photographic results are much more definitive and easier to compare than are audio results. This effect is caused by basic human perceptions that are used to compare and evaluate the final results of audio reproduction and photographic presentation.

In the first case, the ear is the perceiver and the mind the interpreter of the audio result. In the latter the eye is the perceiver and the mind the interpreter of the photographic result. There is a basic difference between these two processes. In the first case a time-sequential comparison is made, and in the second the comparison can be and usually is simultaneous. Because of the time-sequential comparison in the case of audio presentations, judgment is less precise and more easily biased by opinion. One has to jump back and forth between comparisons in the audio case, and this fuzzes up the ability to judge. One cannot hear both presentations at the same time. It is true that A/B or ABX comparisons have been quite successful in ferreting out differences and pinning down differences. But there are still those who choose to believe what they want to, regardless of the truth. One can only hope to show that these persons really can't tell the difference, if any, and show them up for what they are, frauds.

In the case of photographic performance, the comparisons of photographic results are done simultaneously. That is, side by side but at the same time. When simultaneous comparisons are made, the differences, if any, are observed and can be discussed by the viewers while doing the examination. The language of comparison then becomes very precise and the viewers can interact quickly, in real time, to move toward a resolution of differences of opinions. This sort of interaction cannot take place in audio comparisons. Thus differences of opinion often remain unresolved.

I believe that for at least the two reasons stated above, and possibly others, there is a significant difference between the audio and photographic fields, which will continue. Photography will remain relatively free of fakes and frauds, while audio will continue to be replete with them.

Sincerely,
R. A. Greiner
Emeritus Professor of Electrical and Computer Engineering
Madison, WI

(continued on page 34)

By **PETER ACZEL,** Editor

The Ten Biggest Lies in Audio

The punch line of Lincoln's famous bon mot, that you cannot fool all the people all of the time, appears to be just barely applicable to high-end audio. What follows here is an attempt to make it stick.

I strongly suspect that people are more gullible today than they were in my younger years. Back then we didn't put magnets in our shoes, the police didn't use psychics to search for missing persons, and no head of state since Hitler had consulted astrologers. Most of us believed in science without any reservations. When the hi-fi era dawned, engineers like Paul Klipsch, Lincoln Walsh, Stew Hegeman, Dave Hafler, Ed Villchur, and C. G. McProud were our fountainhead of audio information. The untutored tweako/weirdo pundits who don't know the integral of e^x were still in the benighted future.

Don't misunderstand me. In terms of the existing spectrum of knowledge, the audio scene today is clearly ahead of the early years; at one end of the spectrum there are brilliant practitioners who far outshine the founding fathers. At the dark end of that spectrum, however, a new age of ignorance, superstition, and dishonesty holds sway. Why and how that came about has been amply covered in past issues of this publication; here I shall focus on the rogues' gallery of currently proffered mendacities to snare the credulous.

1. The Cable Lie

Logically this is not the lie to start with because cables are accessories, not primary audio components. But it is the hugest, dirtiest, most cynical, most intelligence-insulting and, above all, most fraudulently profitable lie in audio, and therefore must go to the head of the list.

The lie is that high-priced speaker cables and interconnects sound better than the standard, run-of-the-mill (say, Radio Shack) ones. It is a lie that has been exposed, shamed, and refuted over and over again by every genuine authority under the sun, but the tweako audio cultists hate authority and the innocents can't distinguish it from self-serving charlatanry.

The simple truth is that resistance, inductance, and capacitance (R, L, and C) are the only cable parameters that affect performance in the range below radio frequencies. The signal has no idea whether it is being transmitted through cheap or expensive RLC. Yes, you have to pay a little more than rock bottom for decent plugs, shielding, insulation, etc., to avoid reliability problems, and you have to pay attention to resistance in longer connections. In basic electrical performance, however, a nice pair of straightened-out wire coat hangers with the ends scraped is not a whit inferior to a $2000 gee-whiz miracle cable. Nor is 16-gauge lamp cord at 18¢ a foot. Ultrahigh-priced cables are the biggest scam in consumer electronics, and the cowardly surrender of nearly all audio publications to the pressures of the cable marketers is truly depressing to behold.

(For an in-depth examination of fact and fiction in speaker cables and audio interconnects, see Issues No. 16 and No. 17.)

2. The Vacuum-Tube Lie

This lie is also, in a sense, about a peripheral matter, since vacuum tubes are hardly mainstream in the age of silicon. It's an all-pervasive lie, however, in the high-end audio market; just count the tube-equipment ads as a percentage of total ad pages in the typical high-end magazine. Unbelievable! And so is, of course, the claim that vacuum tubes are inherently superior to transistors in audio applications—don't you believe it.

Tubes are great for high-powered RF transmitters and microwave ovens but not, at the turn of the century, for amplifiers, preamps, or (good grief!) digital components like CD and DVD players. What's wrong with tubes? Nothing, really. There's nothing wrong with gold teeth, either, even for upper incisors (that Mideastern grin); it's just that modern dentistry offers more attractive options. Whatever vacuum tubes can do in a piece of audio equipment, solid-state devices can do better, at lower cost, with greater reliability. Even the world's best-designed tube amplifier will have higher distortion than an equally well-designed transistor amplifier and will almost certainly need more servicing (tube replacements, rebiasing, etc.) during its lifetime. (Idiotic designs such as 8-watt single-ended triode amplifiers are of course exempt, by default, from such comparisons since they have no solid-state counterpart.)

As for the "tube sound," there are two possibilities: (1) It's a figment of the deluded audiophile's imagination, or (2) it's a deliberate coloration introduced by the manufacturer to appeal to corrupted tastes, in which case a solid-state design could easily mimic the sound if the designer were perverse enough to want it that way.

Yes, there exist very special situations where a sophisticated designer of hi-fi electronics might consider using a tube (e.g., the RF stage of an FM tuner), but those rare and narrowly qualified exceptions cannot redeem the common, garden-variety lies of the tube marketers, who want you to buy into an obsolete technology.

3. The Antidigital Lie

You have heard this one often, in one form or another. To wit: Digital sound is vastly inferior to analog. Digitized audio is a like a crude newspaper photograph made up of dots. The Nyquist-Shannon sampling theorem is all wet. The 44.1 kHz sampling rate of the compact disc cannot resolve the highest audio frequencies where there are only two or three sampling points. Digital sound, even in the best cases, is hard and edgy. And so on and so forth—all of it, without exception, ignorant drivel or deliberate misrepresentation. Once again, the lie has little bearing on the mainstream, where the digital technology has gained complete acceptance; but in the byways and tributaries of the audio world, in unregenerate high-end audio salons and the listening rooms of various tweako mandarins, it remains the party line.

The most ludicrous manifestation of the antidigital fallacy is the preference for the obsolete LP over the CD. Not the analog master tape over the digital master tape, which remains a semi-respectable controversy, but the clicks, crackles and pops of the vinyl over the digital data pits' background silence, which is a perverse rejection of reality.

Here are the scientific facts any second-year E.E. student can verify for you: Digital audio is bulletproof in a way analog audio never was and never can be. The 0's and l's are inherently incapable of being distorted in the signal path, unlike an analog waveform. Even a sampling rate of 44.1 kHz, the lowest used in today's high-fidelity applications, more than adequately resolves all audio frequencies. It will not cause any loss of information in the audio range—not an iota, not a scintilla. The "how can two sampling points resolve 20 kHz?" argument is an untutored misinterpretation of the Nyquist-Shannon sampling theorem. (Doubters are advised to take an elementary course in digital systems.)

The reason why certain analog recordings sound better than certain digital recordings is that the engineers did a better job with microphone placement, levels, balance, and equalization, or that the recording venue was acoustically superior. Some early digital recordings were indeed hard and edgy, not because they were digital but because the engineers were still thinking analog, compensating for anticipated losses that did not exist. Today's best digital recordings are the best recordings ever made. To be fair, it must be admitted that a state-of the-art analog recording and a state-of-the-art digital recording, at this stage of their respective technologies, will probably be of comparable quality. Even so, the number of Tree-Worshiping Analog Druids is rapidly dwindling in the professional recording world. The digital way is simply the better way.

4. The Listening-Test Lie

Regular readers of this publication know how to refute the various lies invoked by the high-end cultists in opposition to double-blind listening tests at matched levels (ABX testing), but a brief overview is in order here.

The ABX methodology requires device A and device B to be level-matched within +0.1 dB, after which you can listen to fully identified A and fully identified B for as long as you like. If you then think they sound different, you are asked to identify X, which may be either A or B (as determined by a double-blind randomization process). You are allowed to make an A/X or B/X comparison at any time, as many times as you like, to decide whether X=A or X=B. Since sheer guessing will yield the correct answer 50% of the time, a minimum of 12 trials is needed for statistical validity (16 is better, 20 better yet). There is no better way to determine scientifically whether you are just claiming to hear a difference or can actually hear one.

The tweako cultists will tell you that ABX tests are completely invalid. Everybody knows that a Krell sounds better than a Pioneer, so if they are indistinguishable from each other in an ABX test, then the ABX method is all wet—that's their logic. Everybody knows that Joe is taller than Mike, so if they both measure exactly 5 feet 11¼ inches, then there is something wrong with the Stanley tape measure, right?

The standard tweako objections to ABX tests are too much pressure (as in "let's see how well you really hear"), too little time (as in "get on with it, we need to do 16 trials"), too many devices inserted in the signal path (viz., relays, switches, attenuators, etc.), and of course assorted psychobabble on the subject of aural perception. None of that amounts to anything more than a red herring, of one flavor or another, to divert attention from the basics of controlled testing. The truth is that you can perform an ABX test all by yourself without any pressure from other participants, that you can take as much time as wish (how about 16 trials over 16 weeks?), and that you can verify the transparency of the inserted control devices with a straight-wire bypass. The objections are totally bogus and hypocritical.

Here's how you smoke out a lying, weaseling, obfuscating anti-ABX hypocrite. Ask him if he believes in any kind of A/B testing at all. He will probably say yes. Then ask him what *special insights* he gains by (1) not matching levels and (2) peeking at the nameplates. Watch him squirm and fume.

5. The Feedback Lie

Negative feedback, in an amplifier or preamplifier, is *baaaad.* No feedback at all is *gooood.* So goes this widely invoked untruth.

The fact is that negative feedback is one of the most useful tools available to the circuit designer. It reduces distortion and increases stability. Only in the Bronze Age of solid-state amplifier design, back in the late '60s and early '70s, was feedback applied so recklessly and indiscriminately by certain practitioners that the circuit could get into various kinds of trouble. That was the origin of the no-feedback fetish. In the early '80s a number of seminal papers by Edward Cherry (Australia) and Robert Cordell (USA) made it clear, beyond the shadow of a doubt, that negative feedback is totally benign as long as certain basic guidelines are strictly observed. Enough time has elapsed since then for that truth to sink in. Today's no-feedback dogmatists are either dishonest or ignorant.

6. The Burn-In Lie

This widely reiterated piece of B.S. would have you believe that audio electronics, and even cables, will "sound better" after a burn-in period of days or weeks or months (yes, months). Pure garbage. Capacitors will "form" in a matter of seconds after power-on. Bias will stabilize in a matter of minutes (and shouldn't be all that critical in well-designed equipment, to begin with). There is absolutely no difference in performance between a correctly designed amplifier's (or preamp's or CD player's) first-hour and 1000th-hour performance. As for cables, yecch... We're dealing with audiophile voodoo here rather than science. (See also the Duo-Tech review in Issue No. 19, page 36.)

Loudspeakers, however, may require a break-in period of a few hours, perhaps even a day or two, before reaching optimum performance. That's because they are mechanical devices with moving parts under stress that need to settle in. (The same is true of reciprocating engines and firearms.) That doesn't mean a good loudspeaker won't "sound good" right out of the box, any more than a new car with 10 miles on it won't be good to drive.

7. The Biwiring Lie

Even fairly sophisticated audiophiles fall for this hocus-pocus. What's more, loudspeaker manufacturers participate in the sham when they tell you that those two pairs of terminals on the back of the speaker are for biwiring as well as biamping. Some of the most highly respected names in loudspeakers are guilty of this hypocritical genuflection to the tweako sacraments—they are in effect surrendering to the "realities" of the market.

The truth is that biamping makes sense in certain cases, even with a passive crossover, but biwiring is pure voodoo. If you move one pair of speaker wires to the same terminals where the other pair is connected, absolutely nothing changes electrically. The law of physics that says so is called the superposition principle. In terms of electronics, the superposition theorem states that any number of voltages applied simultaneously to a linear network will result in a current which is the exact sum of the currents that would result if the voltages were applied individually. The audio salesman or 'phile who can prove the contrary will be an instant candidate for some truly major scientific prizes and academic honors. At the same time it is only fair to point out that biwiring does no harm. It just doesn't do anything. Like magnets in your shoes.

8. The Power Conditioner Lie

Just about all that needs to be said on this subject has been said by Bryston in their owner's manuals:

"All Bryston amplifiers contain high-quality, dedicated circuitry in the power supplies to reject RF, line spikes and other power-line problems. Bryston power amplifiers do not require specialized power line conditioners. Plug the amplifier directly into its own wall socket."

What they don't say is that the same is true, more or less, of all well-designed amplifiers. They may not all be the Brystons' equal in regulation and PSRR, but if they are any good they can be plugged directly into a wall socket. If you can afford a fancy power conditioner you can also afford a well-designed amplifier, in which case you don't need the fancy power conditioner. It will do absolutely nothing for you. (Please note that we aren't talking about surge-protected power strips for computer equipment. They cost a lot less than a Tice Audio magic box, and computers with their peripherals are electrically more vulnerable than decent audio equipment.)

The biggest and stupidest lie of them all on the subject of "clean" power is that you need a specially designed high-priced line cord to obtain the best possible sound. Any line cord rated to handle domestic ac voltages and currents will perform like any other. Ultra-high-end line cords are a fraud. Your audio circuits don't know, and don't care, what's on the ac side of the power transformer. All they're interested in is the dc voltages they need. Think about it. Does your car care about the hose you filled the tank with?

9. The CD Treatment Lie

This goes back to the vinyl days, when treating the LP surface with various magic liquids and sprays sometimes (but far from always) resulted in improved playback, especially when the pressing process left some residue in the grooves. Commercial logic then brought forth, in the 1980s and '90s, similarly magical products for the treatment of CDs. The trouble is that the only thing a CD has in common with an LP is that it has a surface you can put gunk on. The CD surface, however, is very different. Its tiny indentations do not correspond to analog waveforms but merely carry a numerical code made up of 0's and l's. Those 0's and l's cannot be made "better" (or "worse," for that matter) the way the undulations of an LP groove can sometimes be made more smoothly trackable. They are read as either 0's or l's, and that's that. You might as well polish a quarter to a high shine so the cashier won't mistake it for a dime.

Just say no to CD treatments, from green markers to spray-ons and rub-ons. The idiophiles who claim to hear the improvement can never, never identify the treated CD blind. (Needless to say, all of the above also goes for DVDs.)

10. The Golden Ear Lie

This is the catchall lie that should perhaps go to the head of the list as No. 1 but will also do nicely as a wrap-up. The Golden Ears want you to believe that their hearing is so keen, so exquisite, that they can hear tiny nuances of reproduced sound too elusive for the rest of us. Absolutely not true. Anyone without actual hearing impairment can hear what they hear, but only those with training and experience know what to make of it, how to interpret it.

Thus, if a loudspeaker has a huge dip at 3 kHz, it will not sound like one with flat response to any ear, golden or tin, but only the experienced ear will quickly identify the problem. It's like an automobile mechanic listening to engine sounds and knowing almost instantly what's wrong. His hearing is no keener than yours; he just knows what to listen for. You could do it too if you had dealt with as many engines as he has.

Now here comes the really bad part. The self-appointed Golden Ears—tweako subjective reviewers, high-end audio-salon salesmen, audio-club ringleaders, etc.—often use their falsely assumed superior hearing to intimidate you. "Can't you hear that?" they say when comparing two amplifiers. You are supposed to hear huge differences between the two when in reality there are none—the GE's can't hear it either; they just say they do, relying on your acceptance of their GE status. Bad scene.

The best defense against the Golden Ear lie is of course the double-blind ABX test (see No. 4 above). That separates those who claim to hear something from those who really do. It is amazing how few, if any, GE's are left in the room once the ABX results are tallied.

There are of course more Big Lies in audio than these ten, but let's save a few for another time. Besides, it's not really the audio industry that should be blamed but our crazy consumer culture coupled with the widespread acceptance of voodoo science. The audio industry, specifically the high-end sector, is merely responding to the prevailing climate. In the end, every culture gets exactly what it deserves. TAC

By Peter Aczel, Editor

Four Speaker Systems, Ranging from the Most Ambitious to the Most Ingenious

Let me digress briefly before addressing my intended main themes in the individual reviews. I am always worried that new readers of this publication might not be aware of the dominant role of the loudspeaker in any audio system. Upgrading your speakers has the potential to change your audio life, to take you into a new world of sound; upgrading your electronics will not have anywhere near the same effect—if any.

I dwelled on this subject at some length in Issue No. 25 (see pp. 15-16); here I only want to remind you that your money is more wisely spent on new speakers than on any other audio component. That does not mean I endorse loudspeaker systems in the $20K-and-up category (which extends to $100K and more). The vast majority of those insanely high-priced speakers aren't worth 25 cents on the dollar; many of them are just plain rip-offs. On the other hand, you shouldn't expect the $5000 kind of sound out of $500-a-pair loudspeakers. Truly good speakers are never cheap. That becomes even more of an issue with multichannel home-theater systems. (Again, see Issue No. 25, pp. 16-17.)

My highest recommendation to those who are willing to spend serious bucks on a speaker system remains the Canadian **Waveform Mach** 17, now $8495 the pair direct from the factory. So far I have not found its equal in transparency and lack of coloration. Yes, you need a 6-channel power amplifier to use a pair of Mach 17's with their dedicated 3-way electronic crossover, and that raises the cost considerably, but then you have something you can live with for years without the urge to upgrade. Of course, something in the $1500 to $2000 range will also get you excellent speaker performance if you shop wisely (as our regular readers presumably do); just don't expect the highest degree of refinement.

While I am digressing I should also mention that some time ago I auditioned a preproduction version of the new **Infinity "Interlude" IL40** floor-standing 3-way system at only $998 the pair and was amazed by the undeniably "high-end" sound. It was not on my own turf and far from a complete laboratory test, so this does not constitute a recommendation until they send me review samples. Even so, you should be aware of a whole new family of **Infinity** and **JBL** speakers (both are Harman International brands) representing the long-awaited fruition of Floyd Toole's guidelines. (See Issue No. 24, p. 13.) The speakers range from quite inexpensive to very-but-not-illogically expensive and show some promise of taking the performance-per-dollar index to a new level, mainly as a result of a proprietary diaphragm technology (aluminum sandwiched between two layers of ceramic). Most of the models are just beginning to show up in the stores as I write this, so it's still a waiting game.

In general, new chemistry (i.e., materials science) appears to result in more immediate improvements in loudspeaker design than new physics (i.e., exotic transducer principles). The good old moving-coil driver with a better diaphragm looks like the way to go for a while longer. Having said that, I still want to call your attention to a very interesting transducer development, the distributed-mode loudspeaker (DML) pioneered by NXT, a U.K.-based outfit with serious technological and financial resources (i.e., not a basement operation run by tweaks).

The DML is simply a flat panel, of almost any desired size but very stiff, with a complex bending behavior in response to electroacoustic excitation. It produces sound by breaking up into a large number of seemingly randomized vibrational modes over its entire surface. In other words, it is just the opposite of the perfect piston, resonating in many segments and totally lacking coherence. The amazing thing is that it measures flat and sounds quite accurate. It would appear that a few resonances are bad but lots of random resonances are good. That coherence is under most circumstances a nonissue has been explained to our readers a number of times. The DML is a genuinely different approach to transducer design which would need too many pages here to be explained completely; furthermore, its current implementations are all non-hi-fi and thus not really grist for our mill. There exists the promise, however, of future hi-fi applications, and I find the promise credible; an experimental car stereo system with flush DML panels in the upper dashboard sounded just great to me in a recent demonstration, fully competitive with high-end installations of conventional design. The DML is definitely something to be aware of as the art progresses.

As for the individual reviews that follow, you know the old boxing adage that a good big one will always beat a good little one—but the true aficionado judges each contender in the context of the competition. We have a varied assortment of good/big and good/little here, but is there a weight-division champion in the bunch? I think there is at least one, but you will have to decide after having digested the facts.

AVMS AV-1 TruSonic

Audio Video Multimedia Solutions, 17 Saddleback Court, O'Fallon, MO 63366. Voice and Fax: (636) 978-8173. E-mail: tonyscimemiAVMS@worldnet.att.net. AV-1 TruSonic minimonitor/satellite, $900.00 the pair. Tested samples on loan from manufacturer.

What we have here is the main building block of a complete surround system, used for the front left/right as well as the rear left/right channels. The center-channel speaker (AV-C TruSonic, $750.00) is not reviewed here because it is essentially the same speaker with dual woofers. (Besides, we are planning a comprehensive center-channel survey in an upcoming issue.) Nor is the powered subwoofer AVMS sent me reviewed here because it is not the final version that will be sold with the system. The AV-1 is of course the unit on which the overall quality of the 5.1 (or 5.2) system depends.

For the money, and then some, this is a very nicely built little speaker. My review samples came in black oak veneer and appear very professionally finished. All edges and corners are rounded, albeit with a small radius. The back and the bottom are also veneered. The driver complement consists of a 5½-inch woofer and a 1-inch

dome tweeter. The woofer, with composite paper cone (arguably still the best material for large diaphragms), phase plug, and polymer chassis, is mounted above the tweeter. The enclosure is vented to the rear. The silk dome of the tweeter is slightly recessed in a shallow hornlike cavity. The crossover network is second-order.

My quasi-anechoic (MLS) measurements yielded very nice frequency response curves over a large solid angle. The small separation between the two drivers and the fairly seamless crossover made it quite uncritical whether the calibrated microphone was aimed at the woofer or the tweeter, or halfway between the two—the results were almost identical. On the axis of the speaker the tweeter response appeared to be slightly elevated in the top octave, especially since the two octaves from 2 to 8 kHz are extremely flat: ±1.25 dB. The 8 to 16 kHz octave averages 3 to 4 dB above that reference level, with a well-damped peak at 13 kHz. Now here's the most interesting part: at 45° off axis (horizontally) the elevated top octave falls into line, more or less, with the two octaves below, so that the overall response is actually flatter than on axis, with the exception that the curve plummets above 13 kHz. This behavior indicates good power response into the room and relative flexibility in the choice of listening positions and left-right separation. The phase response is well-behaved at all measurement angles

The bass response does not go very low, as the speaker is designed to work in conjunction with a subwoofer. The vented box is tuned to approximately 56 Hz; the maximum output from the vent is at about 64 Hz. The summed response of woofer and vent is essentially flat down to an f_3 (-3 dB point) of 60 Hz, exactly as given in the specs. Below the f_3 the response rolls off at the rate of 18 dB per octave (QB3 alignment, most likely). Everything appears to be very simple and straightforward. The impedance of the system varies from 6.2Ω. to 23Ω in magnitude and between ±35° in phase, not a difficult load for the amplifier.

Tweeter distortion is negligible, as it nearly always is, but the woofer is unhappy with high-level inputs below the f_3, not surprisingly. For example, a 50 Hz tone at a 1-meter SPL of 90 dB bristles with both even and odd harmonics. The second harmonic (100 Hz) is at the -26 dB (5%) level, the third and fourth at -31 dB (2.8%) each, and it doesn't stop there. At the same SPL, however, any fundamental above 125 Hz is quite clean, with a THD in the neighborhood of -46 dB (0.5%). I call that very acceptable in a 5½-inch woofer of nonexotic design.

The sound quality of the AV-1 exceeded my expectations. Subjectively, I found the speaker to make a better sonic impression when inserted into my home theater system than any number of more expensive units that had resided there before. Definition, balance, and ease of dynamics appeared to improve. On music, in my reference stereo system, the AV-1 did not quite have the airy transparency and exquisite detail of the finest speakers but certainly held its own against anything costing $450 per side and then some. There is nothing really faulty or unnatural about its sound. Definitely recommended.

EgglestonWorks "Isabel"

EgglestonWorks Loudspeaker Company, 435 South Front Street, Memphis, TN 38103. Voice: (901) 525-1100 or (877) 344-5378. Fax: (901) 525-1050. E-mail: ewgroup@ix.netcom. com. Web: www.eggworks.com. Isabel 2-way compact loudspeaker system, $2900.00 the pair. Matching stand, $500.00 the pair. Tested samples on loan from manufacturer.

As soon as I started unpacking the Isabels I became aware that I wasn't in Kansas anymore but in high-end tweako country.

Inside the shipping carton, this obelisk-shaped little speaker is spirally wrapped like a mummy—yards and yards and yards of clingy wrapping to protect the high-gloss black finish. Unlike the usual protective bag or sock, the mummy wrapping is destroyed in the unpacking process. But that's not all. I said "little speaker," but this 2-way compact monitor weighs 55 pounds. Yes, with granite side panels, I kid you not. And *that's* not all. The boxy base the speaker needs to be mounted on is the ultimate embodiment of the high-end audiophile creed of redemption through suffering. The prescribed mounting procedure requires eleven—count them!—steps. The four bolts that are supposed to fasten the speaker to the base must be tightened with various washers from inside the base cavity. To do that successfully one must be either a midget who can crawl all the way into the hollow base or an orangutan with arms twice the length of mine.

Get the picture? No you don't. You are then supposed to fill the base with sand or lead shot through a special fill hole (no water or "any other liquid," we are warned) and screw spikes into the bottom. I could go on but I don't want to create the impression that I developed a cultural antagonism to Bill Eggleston's product before I even tested it. No, I gave it every chance; it's just that I come from another world and my jaw tends to drop when I find myself in the high-end fantasists' Land of Oz. I must hasten to add, for the record, that I callously disregarded the instructions and simply placed the speaker without bolts on top of the unfilled and unspiked base, where it remained anchored by its own weight, solid as a rock. I am sure the rhythm-and-pace suffered hugely as a result, but that was of no consequence to an ignorant and insensitive tin ear like me.

The basic engineering design of the Isabel is, on the other hand, extremely simple. (Let's face it, highly sophisticated electroacoustical engineering seldom goes hand in hand with lead-shot filling.) The granite-reinforced MDF enclosure—hernia city, as I said—is vented to the rear because the front panel is barely large enough for the 1-inch tweeter and 6-inch midrange/bass driver. The tweeter is Dynaudio's Esotar cloth-dome model; the 6-incher is from Israel (Morel), featuring a big motor with double magnet and 3-inch voice coil. Expensive drivers, that's for sure. The internal wiring is supplied by Transparent Audio—most probably high-end fantasy cable of no special electrical advantage.

The dead giveaway of the high-end tweako culture is the crossover. The mid/bass driver is driven naked, directly connected to the amplifier. The tweeter is driven through a single series capacitor, in conjunction with a two-resistor L-pad. The theory is that the simplest possible network will yield the best possible sound—the purest solution and all that jazz. Unfortunately, it doesn't quite work that way. Conceptually, a perfectly controlled, smooth midrange rolloff without a network could be modeled as a lowpass filter that sums to unity with a correctly calculated highpass filter for the tweeter. That involves a lot of fancy math not in evidence in the Isabel. The Morel driver doesn't conveniently roll off at 6 dB per octave as claimed by Eggleston—no mass-controlled diaphragm does—and thus cannot form a perfect first-order crossover with the series capacitor on the tweeter. (Not that a first-order crossover is ideal in any event, but we don't need an argument about religion here.) Bottom line: the 6-inch driver runs out of steam around 4 kHz and the tweeter just sort of backs into the dip there without a really good fit.

My quasi-anechoic (MLS) measurements yielded basically good results in the top two octaves of the audio spectrum and not so good further down. In other words, the Esotar tweeter delivers but the Morel mid/bass and the crossover show their shortcomings. The response from 6 to 17 kHz is flat within ± 1 dB and holds up very nicely even 45° off axis, both horizontally and vertically. The resonant peak of the dome is around 16 kHz. The 6-inch driver has a roller coaster response varying at least ±3 dB and in some cases, depending on how the microphone is aimed, as much as ±4 dB. Not very impressive—and conducive to doubts about a 3-inch voice coil for a 6-inch transducer. Bass response is naturally not very deep with the vented box tuned to 100 Hz and maximum output from the vent at around 130 Hz. Eggleston claims —3 dB at 60 Hz; I don't know how they figured that but I'll let it be. In any case, a subwoofer is indicated for full-range response.

Tweeter distortion is very low, as it nearly always is; at a 1-meter SPL of 90 dB, normalized to 7 kHz, it remains between 0.05% and 0.18% over more than two octaves. The mid/bass distortion at the same SPL, normalized to 500 Hz, is in the 0.16% to 0.56% range down to about 110 Hz, rising rapidly to 10% at 32 Hz. All in all, these are very respectable figures. Impedance, above the tuned box range, varies from 6.5Ω to 12Ω in magnitude and from -24° to +18° in phase, a very easy load for the amplifier.

Despite my essentially negative reaction to the basic gestalt of the Isabel, I am not about to characterize its sound as bad—far from it. If you are used to mediocre speakers, the Isabel will sound absolutely gorgeous to you. Its excellent tweeter makes the all-important upper midrange and lower treble sound sweet, smooth, musical, and nonfatiguing at all levels. If, on the other hand, you are used to the finest speakers, as I am and my associates are, the Isabel won't quite make the grade. There is something lacking in transparency, definition of detail, and rendition of space when judged against the best. For example, the **JosephAudio RM7si "Signature"** (see Issue No. 25) is superior in all those respects at a little more than half the price (exactly half if you count the Isabel's stands). It's a classic case of creative engineering versus high-end chic. The money at Eggleston went into image-oriented attributes, at JosephAudio into performance essentials.

I must admit, however, that when the Isabels are sitting there on their stands, gleaming in high-gloss black from obelisk peak to floor, they do have that "Hey, what's that cool setup you have there?" quality—if that's what you're looking for.

Monsoon MM-1000

Sonigistix Incorporated, 101 South Spring Street, Suite 230, Little Rock, AR 72201. Voice: (877) PC AUDIO (toll free). Web: www.monsoonpower.com. MM-1000 multimedia speaker system, $229.00. Tested samples on loan from manufacturer.

Some time ago I received a phone call from David Clark, the designer of this speaker system for computers. (His company is DLC Design; Sonigistix is one of his ancillary enterprises.) He told me he would send me the $229 Monsoon MM-1000 for testing and asked me to judge its nearfield sound, when properly deployed in a desktop computer setup, as if I were evaluating a cost-no-object home-audio model from my normal listening position.

The man has *cojones,* I said to myself. But wait! It turned out he was basically right. I am not saying that my Waveform Mach 17 reference speakers have been equaled or bested. I am saying that the sound of the Monsoon, as experienced with my ears about 18 inches from the satellites, is of the highest fidelity and not an obvious comedown after listening to any high-end speaker in a conventional setup. More about that below.

The MM-1000 system consists of three pieces: two slim panels, each only slightly larger than a business-size envelope, and a small cube, less than a foot in each dimension. The panels are planar magnetic transducers—mini-Magneplanars so to speak—and the cube houses a 5¼-inch woofer plus all the electronics and controls. The box is tuned to 53 Hz (manufacturer's spec). There's also a tiny hard-wired remote volume/mute control. The electronics include two 12.5-watt amplifiers for the panel speakers, a 25-watt amplifier for the woofer, and a 200 Hz third-order active crossover. The controls on the woofer enclosure adjust overall bass level, volume, and on/off 6 dB bass boost at 55 Hz. That's a lot of stuff for $229 at retail, even if none of it qualifies as "audiophile" grade. Sonigistix's parts buyer must be quite resourceful. (Incidentally, the Monsoon MM-1000 is included in some Micron Millennia computer packages.)

My standard methods of loudspeaker measurement are not relevant to this type of system, which is intended to operate in a confined desktop environment with hardly any distance between the transducers and the listener. The 1-meter quasi-anechoic (MLS) response of an individual MM-1000 planar magnetic satellite shows a steady decline of approximately 6 dB per octave throughout its range, and that's certainly not what the ear perceives at the intended listening distance with a reflective desktop interposed. David Clark is one of the grand masters of car-sound engineering, and it appears that the same sort of perceptual response massaging took place here as is required for the special acoustics of an automobile interior. The woofer, on the other hand, is a straightforward vented-box design with essentially flat response down to an f_3 (-3 dB point) of 52 Hz, according to my nearfield measurement of the summed driver and vent. Maybe that's what the specs mean by "tuned to 53 Hz" because the null in the output of the driver—what I call the "tuned to" frequency—is 47 Hz in my sample. Small quibble—52, 53, 47, whatever—it isn't 27 and it can't be. It's just a very nice small woofer. Distortion is quite low; at any frequency above f_3 and any SPL even momentarily tolerable at the normal listening position, THD remains below 1.5%, in most cases well below. The planar satellites stay below 0.5% THD at any frequency within their range and any SPL that the 12.5-watt amplifiers can sustain (at some point buzzing ensues, but only on steady-state signals, not on music). Tone bursts of any frequency are reproduced without ringing. Thus, the system can be declared to have some pretty decent measurable performance characteristics, even if one disregards the price—but is that what makes it sound good? I'm sure that's part of it, but the careful tailoring of the satellite panels' response to the specific listening environment is probably the most important factor. I can't be certain.

I'm on much firmer ground when I tell you what I hear when I insert a well-recorded music CD into my computer's CD/DVD tray and set the volume to the subjectively most convincing level. Are there any obvious colorations? No. Are all the instruments and voices natural and clear? Yes. Are the three-dimensional characteristics of the recording space and the deployment of the performers audible? Definitely. Are the dynamics restricted? Not at all. If the recorded sound is especially beautiful—shimmering strings, aerated woodwinds, golden brasses—does that special thrilling quality come through? It does. Quite an amazing product.

Even so, don't misunderstand me. The Monsoon MM-1000 is not the $229 solution to the problem of finding a reasonably priced reference-quality speaker system for your listening room. You will not like it if you insert it into your regular stereo system and listen to it from your favorite armchair. It is designed for a highly specialized application, just like a pair of headphones. In that application, I believe it will satisfy the most demanding users. What I admire about it especially is that it is such an elegant piece of engineering, in the true sense of the word. Engineering, to me, means the straightest line to the simplest correct solution. You think a $156,000 Wilson Audio speaker system is engineering? No, it's an undisciplined exercise in excess, a Caligula's feast. The Monsoon MM-1000 is engineering.

Revel "Salon"

Revel Corporation, a Harman International Company, 8500 Balboa Boulevard, Northridge, CA 91329. Voice: (818) 830-8777. Fax: (818)892-4960.
E-mail: support@revelspeakers.com. Web: www.revelspeakers.com. "Salon" floor-standing 4-way loudspeaker system, $14,400.00 or $15,500.00 the pair, depending on finish. Tested samples on loan from manufacturer.

This is a tough one. Kevin Voecks, Snell's former ace and now chief designer of Harman International's ultra-high-end Revel division, had told me before the Revel "Salon" made its debut that it would be "the world's best speaker." I always had, and still have, the highest respect for Kevin's work but I can't quite go that far in my ranking of the Salon. Indeed, I am still inclined to regard the aging Snell Acoustics Type A as Kevin's masterpiece. Not that the Salon is anything less than a very fine loudspeaker system, exemplifying some of today's most sophisticated design approaches. There are a few things about it, however, that I like a lot less than I expected to.

To begin with, the Salon is unnecessarily awkward physically. Each speaker system in its shipping carton weighs 240 pounds. I am pretty ingenious when it comes to moving huge packages around without lifting them (pushing on a dolly, sliding on a carpet, tumbling the monsters end over end, etc.), but this one gave me a terrible time. The speaker incorporates a separate main enclosure, a separate tweeter/midrange enclosure, and two huge detachable side panels, so there is really no reason other than economy (or support of hernia surgeons) to ship it all screwed together; it could be neatly broken down into manageable modules. It's a no-brainer. That the dealer or the end user (you or I) couldn't assemble the modules as solidly as the factory is a tweako high-end bugaboo—and not the only one I discerned in the Salon. Another is wire fastener/connector phobia (i.e., solder fetishism), which I discovered when I had to replace both tweeters, front and rear, in my left channel. A momentarily interrupted ground connection somewhere in my A/B lash-up had blown both units, raising the question of excessive fragility (but that's not my point here) and necessitating the installation of new tweeters hastily obtained from Revel. I was genuinely disturbed to see that Revel tack-solders the fat wires from the crossover network to the tiny terminals of the tweeters without any mechanical connection. I have seen some very high-quality snap-on connectors (e.g., in JBL speakers and others) that make life a lot more pleasant should servicing be required—and, no, they don't fail; they don't introduce more than a hundredth of an ohm; they just cost more than a blob of solder.

There, I'm already grumpy and I haven't even started to discuss the sound or the measurements. Yes, I also have some good things to say, but let's begin at the beginning.

The Salon is designed with seven drivers per side, five of them created from scratch in Harman International's facilities. The two exceptions are the 1.1-inch aluminum-dome front tweeter and the 0.75-inch aluminum-dome rear tweeter, which are imports (the front unit a very expensive one from Scan-Speak). The latest in-house tweeters were apparently not yet available when the Salon was in the development stage. Not so the midrange driver, a unique in-house design with a 4-inch (!) concave titanium dome—very impressive. The midrange and the front tweeter are housed in a separate enclosure with thickly rounded edges and corners to control diffraction. The woofer complement consists of three vertically deployed 8-inch drivers in the main enclosure, which is loaded with a huge flared port firing rearward. On top of the woofers there is a 6½-inch midbass driver, and the little rear tweeter is mounted near the top of the main enclosure's back. The three woofers and the midbass all have concave diaphragms made of an apparently very high-tech polymer material. The crossover network uses 24-dB-per-octave slopes (naturally), air-core inductors, and film capacitors; the crossover frequencies are 125 Hz, 450 Hz, and 2.2 kHz. Two pairs of binding posts in the rear provide the usual choice of single wiring, biwiring, and biamping; a level control for the front tweeter provides —1, -0.5, 0, +0.5, +1 settings; a level control for the rear tweeter can be set to Off, 0, or -3; a continuously variable LF compensation control has a range of—2 dB to +2 dB centering on 50 Hz.

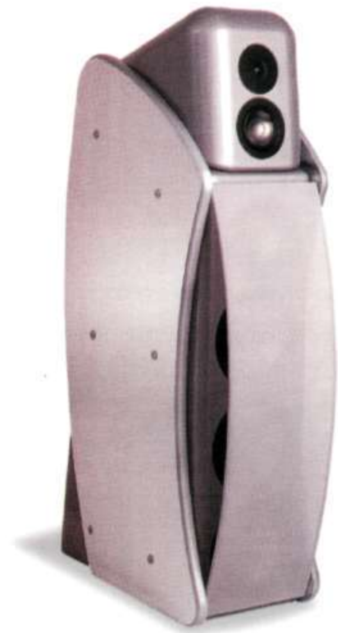
The most controversial thing about the Revel Salon right out of the shipping carton is its appearance. Are the gigantic kidney-shaped side panels functional or an over-the-top "contemporary design" conceit? They do add mass and stiffness to the cabinet, but the speaker can function without them (grilleless, to be sure, as they anchor the strangely bowed-out nondiffractive grille-cloth assembly). Side panels of several other colors (my samples came in rosewood) can be substituted, as can grille cloths (mine were dark gray), and the enclosures themselves can be ordered in a number of finishes (mine were high-gloss black). Thus the basic gee-whiz contempo look comes in gradations from relatively conservative, such as my samples, to a bold color combination approximating the Polish flag. First-time reactions to Revel's visual statement range from bravo to yuck. Mine was quite favorable, on the whole. I think the industrial designer's marching orders were to maintain a family look in all Revel models, regardless of size. That's not easy.

I decided to listen to the Revel Salon before measuring it because I did not want to be influenced by what I presumed would be outstanding measurements. (Not that such a presumption is entirely without influence.) I fired up the speakers with a good orchestral CD through 200 watts per channel and was almost immediately struck by the total absence of dynamic compression. That may be the Salon's strongest feature. Those special drivers are doing the job. The bass is rock-solid and goes low enough, and then some, to obviate subwoofers. What about transparency and lack of coloration? That was a more difficult assessment, requiring further investigation.

Since Floyd Toole sets the general guidelines for all Harman International speaker designs (leaving the actual execution to the individual designers) I followed his well-known and by now axiomatic protocol for A/B comparisons. To wit: the speaker under test and the reference speaker must be compared one on one, mono versus mono, side by side, freestanding, at matched levels. My reference speaker, as already stated above, is the Waveform Mach 17. I used identical monoblock power amps with volume controls to drive the speakers, carefully matching the pink-noise SPLs within a fraction of a dB with a sound-level meter at my listening location. I turned off the rear tweeter of the Salon to make the listening setup as symmetrical as possible. Initially I had all level controls of the Salon at 0.

Please note that this was not a blind test. It would have required a repositioning carousel and a huge acoustically transparent screen to make it blind. Our laboratory is not quite on that level of sophistication. Besides, very few speakers sound so much alike that a blind test is absolutely needed. (Take it from a hard-core ABX advocate when it comes to electronic signal paths.) Above all, be aware that even such an objectively configured listening test is subjective in its conclusions; only measurements are provably objective.

So, what did I hear? With the Salon's tweeter level set to 0, it had a disembodied-top type of coloration to my ears, not even close in neutrality to the Mach 17, which had been optimally balanced through its electronic crossover. The most listenable output of the Salon appeared to be obtainable



with the tweeter level set to -1, but even then the balance from the top down to the lower registers wasn't quite as seamless and natural as that of the Waveform. The midrange of the latter also appeared to be better rounded and somehow more lifelike than the Revel's, particularly on singers' voices, both male and female. Mind you, the difference wasn't huge, but every time I quickly switched from the Revel to the Waveform the sound appeared to open up, smoothen out, and acquire a more credible perspective—subtly, not dramatically. A briefly participating female listener felt that the Revel's sound was slightly irritating (I couldn't quite agree) and the Waveform's natural and nonfatiguing (I agreed). Of course, all of the above could be contradicted by a highly qualified listener whose sonic tastes are different from mine because the sound of the Revel Salon is good enough to be subject to pro/con argument on the highest level. My own perception is that the Salon is very good and the Mach 17 is the best—and the best is the enemy of the good, as Voltaire said.

Interestingly enough, the Snell Type A sounds much more like the Waveform Mach 17 in a similar A/B comparison (see Issue No. 24), hence my aforementioned partiality to the Snell. I am fully aware that the Harman International test facility built under the direction of Floyd Toole is more advanced than his older NRC facility in Canada and that the Revel Salon is the result of an even more sophisticated design protocol than the NRC-derived Snell Type A, but a better tool doesn't necessarily guarantee a better result. I must also add, in all fairness, that the Snell was not available for an A/B/C comparison.

When it came to the measurements I must confess I was seriously intimidated by the engineering pedigree of the speaker. Unlike Kevin Voecks, I can't run accurate response curves at 72 different points in a 4Π space inside a gigantic anechoic chamber—and that's just a small part of their design and test procedures. It is beyond the scope of this review to discuss in detail everything that Revel does; go to www.revelspeakers.com for the ultimate in audio-geek intimidation. They do all sorts of averaging, smoothing, power-response figuring, psychoacoustic massaging, etc., as against my pitifully few 1-meter and 2-meter quasi-anechoic (MLS) curves and ridiculously simple nearfield measurements. All I can say in defense of my clearly less sophisticated methods is that they have served me well in the past to identify strengths and weaknesses and to support my subjective perceptions with valid objective data.

So, I've hemmed and hawed long enough; now I'll have to say it: I was unable to obtain as good measurements on the Revel Salon as on the Waveform Mach 17 (or the Snell Type A for that matter). No matter which driver I aimed the microphone at, from 1 meter or 2 meters, the absolute best on-axis response I could get was ±3.5 dB (i.e., all swings contained within a 7-dB strip). That was taken from 1 meter, on the axis of the tweeter, with the tweeter level set to - 1 . Furthermore, every one of the many response curves I experimented with showed comb-filter squiggles all over the place, something I never see in my routine measurements. What's more, all the curves had a maximum dip in the 3.5 to 4 kHz band, where there isn't even a crossover. I could dismiss these anomalies as measurement artifacts (as I am sure Revel would) if—but only if—the speaker had sounded more neutral to my ears than the Waveform or the Snell. What caused them is subject to speculation, perhaps the spacing of the drivers, perhaps the protruding upper-front corners of the side panels, perhaps something else altogether or a combination of such things. One response curve that was quite impressive, on the other hand, was the one taken 45° off the tweeter axis at a 1-meter distance, with the tweeter level set to 0 (not —1, alas). The tweeter response remained within a 2.8 dB strip up to 9 kHz and dropped only 5 dB at 14 kHz. That indicates very good power response as well as the absence of the head-on anomalies. Mysterious.

Bass response shows the classic B4 alignment, flat down the box-tuning frequency of 24 Hz (—3 dB point), fourth-order slope below that. No problem, no mysteries. The impedance curve of the system is similarly unproblematic, between 3.2 and 9.3 ohms in magnitude from 20 Hz to 20 kHz (6 ohms nominal) and ±33° in phase over the same range. It's a load you could drive with a cheap Pioneer receiver if you are unafraid of the high-end audio Furies. Distortion is outstandingly low at all frequencies; I think even David Hall of Velodyne would approve (or at least not disapprove). For example, the nearfield THD of the woofer at 50 Hz at a 1-meter SPL of 97 dB is 0.5%, rising with a steady slope to 2.5% at 30 Hz. Further up, the midbass/midrange THD hovers around 0.5% up to 1 kHz at a 1-meter SPL of 90 dB. Above that range, where the tweeter begins to take over, the THD is negligible. Once again, no problemo.

So—what kind of recommendation can I make regarding the Revel Salon? If somebody gives it to you for your birthday or Christmas, keep it. There aren't too many better speakers out there. If you are using your own money, you have some options I consider superior, as I have already explained. The Revel people will of course disagree on the grounds that their measurement procedures are more comprehensive and accurate than mine and their listening protocol more objective. That may very well be true but I have one advantage over them: I don't care whose speaker comes out on top, but they do.

Postscript: After I had written the above, I heard a strange explanation of the 3.5 to 4 kHz dip in the Revel Salon's response. The explanation (excuse?) does not appear anywhere in the Revel literature or on their Web site, where all is flatness, sweetness, and light. No, it's what the Revel designers are saying in private discussions, at least according to my admittedly not quite first-hand informants. They (allegedly) say the dip is necessary in mono to compensate for subtle head-geometry effects in stereo. In other words, the response needs to be a little bit "bad" in mono so it will be totally "good" in stereo. I have a couple of problems with that, if it's indeed what they are saying. What happened to Floyd Toole's famous mono listening-test protocol? Why didn't the slight imperfections I heard in mono disappear totally in stereo? Are frequency-response cancellations/reinforcements around my thick head the whole story when I hear small colorations? How come this whole subject hasn't come up in connection with other NRC/Toole-derived loudspeaker designs? Hey, this may be the most sophisticated insight in the world of audio today, but why not come out and tell us all? We're all ears. TAC

By Peter Aczel, Editor
David A. Rich, Ph.D., Technical Editor
Glenn O. Strauss, Contributing Editor

Power Amplifiers and Outboard D/A Converters

The good prevails; the bad and the ugly are falling behind. That's basically the current state of audio electronics, both analog and digital—but you still can't believe every claim.

Regular readers of this publication are familiar with our position on the audio quality of electronic signal paths—what is audible and what is not in valid, controlled listening tests. Newcomers should read a few back issues (check out No. 24 and 25 for openers). We can't keep going over the same ground in every issue.

Here I just want to emphasize once again that in engineering, as in other fields, good thinking costs no more than bad thinking. Good measurements are proof of good thinking; that's why we emphasize them, whether "you can hear the difference" or not. Maybe you can't hear 0.05% THD, but 0.005% is just as easy to achieve with good thinking and a lot more reassuring. Besides, if you cascade four or five of those devices with not-so-great measurements . . . who knows? *—Ed.*

5-Channel Power Amplifier

Bryston 9B ST

Bryston Ltd., P.O. Box 2170, 677 Neal Drive, Peterborough, Ont., Canada K9J 7Y4. Voice: (705) 742-5325 or (800) 632-8217. Fax: (705) 742-0882. Web: www.bryston.ca. 9B ST 5-channel power amplifier, $3695.00. Tested sample on loan from manufacturer.

In amplifier design, Chris Russell and Stuart Taylor are a combination like Joe Montana and Jerry Rice in the NFL of the 1980s—as good as it gets. The basic Bryston power-amp topology (the one that made David Rich call Chris "a ridiculously good engineer" in a long-ago issue of this journal) has changed only slightly over the years. The main improvements in the present line have to do with physical layout and how the gain is shared between the stages, the net result being a lower noise floor. The ST suffix following the model number credits Stuart Taylor for the improvements. I have already reviewed the 3-channel and 4-channel ST amps in the line (see Issue No. 24); the 9B ST is their 5-channel model and the flagship of the line (at least until a higher-powered version now in the pipeline is released).

This is truly a gorgeous piece of equipment. No wonder Bryston likes to exhibit it with the cover off at the various shows. The layout is of the utmost architectural beauty because of its uncluttered simplicity. Five self-contained, independent mono modules are arrayed side by side, each fully operational by itself. Only the line cord and the on/off switch are shared. Each module offers unbalanced, balanced, or high-gain (+6 dB) balanced operation via a 3-position input selector switch. The inputs accept RCA, XLR, and standard phone plugs. The transformers generate no mechanical hum off the chassis, not even the slightest, and there are no on/off thumps through the speakers, ever. The only clue to power on/off is the five-LED front-panel display. The overall impact of the amplifier in use is that you are in totally competent, totally professional hands and nothing untoward can happen. Both the visual and functional aspects of the 9B ST contribute to that impression.

On the lab bench the measurements are equally impressive. This is one of the very few amplifiers yielding identical distortion-versus-output curves at *any* frequency. The 20 kHz curve tracks the 20 Hz and 1 kHz curves so closely that the three, when superimposed, look like one curve. No "dynamic distortion" here! Into 8ft the curves bottom out at -90 dB, into 4Ω at -88 dB. These minima are at the precise clipping point, which is 125 W into 8Ω and 200 W into 4Ω. Needless to say, the distortion is entirely noise-dominated, the ruler-straight curves (is that an oxymoron?) declining 6 dB for every 3 dB increase in power output, starting at -50 dB with 10 mW out into 8Ω. I have seen slightly lower distortion figures but I have never seen greater consistency.

The PowerCube of the 9B ST painted a highly satisfactory picture as regards dynamic power and current limiting. This measurement, which no other American audio magazine performs, tests short-burst power capability into reactive (i.e., real-world) loads. For a detailed explanation, see Issue No. 20, where the test was first introduced. Into resistive (0°) loads of 8Ω/4Ω/2Ω/1Ω, dynamic power of the measured channel was 211W/350W/516W/574W Into capacitive (-60°/-30°) and inductive (+30°/+60°) loads, dynamic power was slightly up at 8ft/4ft/2ft (ideal) and at least not sagging, though not up, at 1Ω (acceptable). Those are good numbers considering the continuous power rating of 120 W per channel into 8Ω.

Crosstalk between adjacent channels at 1 W output into 8Ω also declines 6 dB per octave as the frequency is lowered, starting at -56 dB and -65 dB at 20 kHz in the two channels I measured and ending at -105 dB and -104 dB, respectively, at 20 Hz. Of course, the channel separation would be virtually infinite, were it not for the close proximity of the mono modules. And, yes, I almost forgot: the frequency response, at 1 W into 8ft, is ±0.0 dB up to 5 kHz, declining to -0.07 dB at 20 kHz and -0.46 dB at 50 kHz. Around here, we call that flat.

Bottom line: this is an impeccably designed and constructed 5-channel amplifier, far from cheap but not shamelessly inflated in price, since every dollar is in evidence right there "under the hood." You can undoubtedly have the same sound for less money but not the same satisfaction.

—Peter Aczel

Mono Power Amplifier

Bryston PowerPac 60

Bryston Ltd., P.O. Box 2170, 677 Neal Drive, Peterborough, Ont, Canada K9J 7Y4. Voice: (705) 742-5325 or (800) 632-8217. Fax: (705) 742-0882. Web: www.bryston.ca. PowerPac 60 mono power amplifier module, $450.00. Tested sample on loan from manufacturer.

What we have here is one power amplifier channel (i.e., mono) from the Bryston B-60 integrated stereo amplifier, packaged as a pancake module with mounting flanges. Who needs it? Among others, I do; I use it as a lab bench amplifier for testing speakers. It can also be mounted on the back of a loudspeaker enclosure to save space and eliminate speaker cables (all you need is a pair of jumpers a few inches long). There is also a PowerPac 120 version ($795.00), which is twice as powerful and equivalent to one channel of a -B ST series amplifier. I tested the 60 because it is compact, cute, cuddly, and convenient—and far from underpowered for a good many applications.

A switch on the chassis selects unbalanced or balanced input, and the inputs will accept RCA, XLR, or standard phone plugs. A ground lift switch to counteract ground loops, a Bryston specialty, is also provided. The spacing of the speaker terminals will fit a good old-fashioned double banana plug, thank goodness.

As for the measurements, I'll make a sweeping statement, since I hate to repeat myself. Take all the wattages I reported for the Bryston 9B ST above, reduce them by 40% to 50%, leave the distortion and frequency dB's alone—and you have the whole picture. At its lower power rating, the PowerPac 60 is every bit as perfect, maybe even a hair better. The only exception to that statement is the dynamic power into 1Ω, which does not track the other figures but sags a little bit as the power supply reaches its limit. (You still get 180 W or better into all 1Ω reactive loads.) This is one hell of a little amplifier.

Bryston has given ample proof to the world that good engineering, meaning science without excuses, results in good amplifiers. The ultrahigh-end voodooists with their cockamamie theories and shamelessly inflated prices have absolutely nothing to offer by comparison, and the lower-priced brands have something to measure themselves against.

—Peter Aczel

Outboard D/A Converters

Entech NC205.2/NC203.2

Entech: Entertainment Technologies, a division of The Monster Group, 274 Wattis Way, South San Francisco, CA 94080. Voice: (650) 871-6000. E-mail: entech@monstercable.com. Web: www.monstercable.com. Number Cruncher 205.2 20-bit stereo DAC with 5-pole anti-alias filter, $450.00. Number Cruncher 203.2 20-bit stereo DAC with 3-pole anti-alias filter, $299.00. Tested samples on loan from manufacturer.

These are two rather stylish little aluminum boxes shaped like Quonset huts, the 203.2 not much bigger than the fat modems of a few years ago, the 205.2 twice as deep but with the same front-panel size and cross section. Both come with 16-volt outboard power supplies ("wall warts"). I was really tickled by the contradiction between the unquestionably superior performance of the Number Crunchers and their tweako/weirdo owner's manual. A full month of burn-in is recommended; soundstaging and front-to-back layering are lovingly dwelt on (quotes from Robert Harley, voodoo-science advocate extraordinaire); interconnect cable magic is invoked—you name it. Then, when you look at Entech's affiliation, all is explained. It is a division of The Monster Group, i.e., Noel Lee's empire. What else would you expect from the founding father of the cable cult?

There is a simple reason for the excellent performance of the Number Crunchers. It can be summarized in three words: Crystal Semiconductor Corporation. They make digital chips of highly advanced design, and Entech uses two of them in each NC unit: the CS8412 low-jitter digital input receiver and the CS4329 delta-sigma DAC, specified to yield 20-bit resolution. A good result is virtually guaranteed unless you mess up the circuit—and Entech didn't (thank the favorable conjunction of stars in Noel Lee country). The analog output in both models is via a Burr-Brown OPA2134 chip, with a 5-pole anti-aliasing filter in the 205.2, a 3-pole in the 203.2. A further difference is the number of digital inputs: two coax and one optical on the 205.2, just one of each on the 203.2. The 205.2 also has one more independent voltage regulator than the 203.2—six instead of five—plus a digital-domain phase switcher not present in the 203.2. As a result, the front panel of the 205.2 is a lot busier than that of the 203.2, which sports a single "data locked" light and that's all.

The irony is that the little 203.2 measures even better than the more elaborate 205.2, undoubtedly because of the utterly simple, straight-through Crystal/BB signal path without add-ons. Indeed, I have measured very few, if any, D/A converters that equaled it in every respect. Distortion at full-scale digital input never rises above -94.5 dB at *any* frequency and is down to —97.2 dB at some of the *higher* frequencies. With the digital input reduced to -20 dBFS and the distortion normalized to full scale (thus washing out the gain-related analog distortion component), the result is in the range between -101.5 dB and -103.2 dB at *all* frequencies. Go, Crystal, go! The single-point (1 kHz) noise measurement, as referenced to full-scale output, yielded exactly the same figure in both channels: -102.5 dB. The FFT spectrum of a full-scale 1 kHz tone shows no harmonics of any order above the -94 dB level. A dithered 1 kHz tone at -60 dBFS has a totally blip-free FFT spectrum ("Rob Watts test") and a bin-by-bin noise floor of -134 dB, the best I have ever measured. Gain linearity is totally error-free (0.0 dB error) down to —96 dB and then creeps toward —0.5 dB error at the —110 dB level. Frequency response with 0 dBFS input is ±0.0 dB from 10 Hz to 2 kHz, rolling off to -0.3 dB at 20 kHz. Crosstalk decreases at the rate of 6 dB per octave as the frequency is lowered, starting with -71 dB at 20 kHz and dropping to -120 dB at 25 Hz. Read these numbers and weep, all you makers of ultrahigh-end DACs.

The 205.2 also excels on the lab bench but falls short of the 203.2 in a number of tests. Frequency response, crosstalk, and gain linearity are roughly the same, but the single-point noise is 1 dB worse, full-scale distortion is 2 dB worse, and the -20 dBFS distortion normalized to full scale is roughly 6 to 7 dB worse. The "Rob Watts test" (see above) shows odd-order harmonics up to 19 kHz rising 18 to 28 dB from the bin-by-bin noise floor of -132 dB. These are still very good numbers, but for $101.00 less the MSB Technology "Link" (see below) measures quite comparably and offers so many more features that there is simply no contest.

On the other hand, the Entech NC203.2 at $299.00 has some appeal as a possible enhancement for a single digital source (CD, MD, DAT, whatever) with a less up-to-date internal DAC. For just one additional Ulysses Grant bill, however, there is once again the much more versatile MSB.

—Peter Aczel

Outboard D/A Converter

MSB Technology "Link"

MSB Technology, 14251 Pescadero Road, La Honda, CA 94020. Voice: (650) 747-0400. Fax: (650) 747-0405. Web: www.msbtech.com. The Link 24-bit 96-kHz DAC, $349.00. Tested sample on loan from manufacturer.

When Federal Express delivered this unit, I was amused that the driver was holding the box with two hands. Every other $349 product I've seen lately has been a subcompact, lightweight package. The amusement ended when I was handed the box, which indeed had some heft to it.

And I think this speaks to the heart of The Link—tremendous value for the dollar. A physical examination will bear that out. This digital-to-analog converter is full size at 17" (w.) X 14" (d.) X 1¾" (h.). The chassis and top plate are steel. The front panel has no switches or controls, as the unit is fully automatic in operation, detecting the appropriate sampling rate and digital source (coax or Toslink) and switching accordingly. Eight LEDs display power, input source, and sampling frequency (32, 44.1, 48, 88.2, and 96 kHz). The rear panel has a DIN-style power connection and very heavy-duty RCA jacks, heavily gold-plated. The only obvious concession to price in the physical plant is a rather pedestrian metal flake paint job.

The power supply further testifies to the high-value presentation. Rather than a cheesy "wall wart," MSB provides a real power source, a separate encased unit with multiple supplies for the digital and analog circuits (+8 V and ±15 V accordingly). Inside, the theme continues. There is one large, quality dual-layer circuit board measuring 13" X 6", leaving room for possible functional upgrades down the road. There is no point-to-point wiring in evidence, simply a ribbon connector to the front-panel LED display. The board layout is very tidy, and there are no evident tweak design proclivities such as "plutonium lollapalooza" capacitors. Thus, MSB is free to offer proper engineering without silly price inflation. The circuit-board power station has four independently regulated supplies filtered with 3300 µF capacitors—two feed the digital circuitry, two others the analog.

The digital circuit is designed with a Crystal CS8414 receiver chip, which supports sampling rates up to 100 kHz, and a Burr-Brown PCM1716 DAC. The latter is a multilevel delta-sigma processor, specified to decode digital signals up to 96 kHz with 24-bit resolution, coupled with a digital filter using 8x oversampling at 96 kHz. According to Burr-Brown, the DAC architecture "improves audio dynamic performance and reduces jitter sensitivity in actual applications." The 1 kHz distortion spec is a low -96 dB at 0 dBFS. MSB uses the Motorola analog output op-amps' direct-coupling capability and servo monitors to keep things safe. Relay switching in The Link is via a quality Siemens unit.

Measurements in the laboratory of ***The Audio Critic*** could not quite duplicate the full-scale distortion spec of -96 dB at 1 kHz. The reading obtained was -94.3 dB, with rising distortion at higher frequencies, reaching a maximum of—86.3 dB in the 5 kHz to 7 kHz band. That this is essentially gain-related analog distortion was clearly indicated by the greatly improved figures as the digital input was reduced to -20 dBFS and below. MSB's noise figure of-106 dB was not successfully duplicated, either; the best measurement was -102.8 dB, which of course is still an excellent result. The FFT spectrum of a dithered 1 kHz tone at -60 dBFS ("Rob Watts test") was clean as a whistle, with a bin-by-bin noise floor of-130 dB (unusually good). Crosstalk was totally negligible, ranging from -101.5 dB at 20 kHz to between -118 dB and -132.5 dB anywhere below 1 kHz. Frequency response with 0 dBFS input was tipped up 0.3 dB at 10 Hz and rolled off 0.5 dB at 20 kHz but remained within ±0.1 dB from 20 Hz to 9 kHz. Gain linearity was of the utmost perfection, with only —0.4 dB error at -110 dB. (That's the delta-

Can a Car Radio Outperform a High-End FM Tuner?

Blaupunkt Alaska RDM 168

Blaupunkt-Werke GmbH, distributed in the Americas by Robert Bosch Corporation, Sales Group, 2800 South 25th Avenue, Broadview, IL 60153. Voice: (708) 865-5634. Web: www.blaupunkt.com. Alaska RDM 168 mobile 4-channel receiver with FM/AM tuner, CD player, and wireless remote control, $369.95. Tested sample on loan from manufacturer.

As our readers know, we do not cover car audio, but this is really about FM reception in the home, not on the road. Blaupunkt makes complete car-audio electronics, of which this unit is a representative example (though not the latest). Into a tiny chassis it packs FM, AM, 4-channel preamp, 4-channel power amp, equalizer, CD player, RDS display, clock—I could go on, there's more. Of all these features only the FM tuner is of special interest here, for a reason I'm coming to in a moment. I must state up front, however, that neither David Rich nor Richard Modafferi have had their hands on the Blaupunkt so far; this is merely a first-impression report, to be followed in the next issue by an engineering evaluation featuring the combined expertise of those two worthies. Other FM tuners are scheduled to be included as well. Blaupunkt models keep changing, but their unique FM circuitry remains the same.

Here is what I did: I hooked up the Blaupunkt to a Radio Shack 12-volt 1.75-amp unregulated dc power supply ($29.99) instead of the car battery it is designed for. Then I used an available Blaupunkt adapter to access the unit's 3-volt preamp output and bypass the power amps. I took the FM tuner out of my reference system and plopped the adapted Blaupunkt in its place. Everything else remained the same, including the excellent **Terk FM Pro FM-50** indoor/outdoor powered antenna (see Issue No. 25), which I had already used with more than a few high-quality FM tuners. (Connecting the antenna required one Blaupunkt and two Radio Shack plug/jack adapters—a real kludge, but this was an experiment, not a neat installation.)

Now then, why did I go to all this trouble? Because, for the first time in the world, Blaupunkt has FM circuits that process the IF signal digitally. The dirty little secret of FM tuner design is that mobile radio engineers have to be much more savvy and creative than their high-end home-audio counterparts—because the chalsigma advantage.)

In actual use, The Link was tried with two fairly low-priced DVD players. One was the Pioneer DV-414 ($525.00), which can output a 96 kHz PCM bit stream to test The Link's ability to play the high-resolution 96/24 CDs, such as those from Chesky and Classic Records. It locked on with no problems, and the perceived sound was exceptionally transparent. The other player was a Toshiba SD2108 ($599.00).

Even though most quality CD players have exemplary performance, some of the inexpensive DVD players have weak analog sections and cheap digital processors. Here, a better-performing unit such as The Link can be a step up. Another possible use of an outboard DAC such as The Link is in installations with multiple digital sources at different sampling frequencies. You will still need a digital switching device such as Entech's, or MSB's own Digital Director, but be assured the D/A conversion will be competently accomplished by The Link. The Link also acts as a nice base for your player, giving a finished look to your digital sound stream. *(Just make sure the legs of the player allow some air space above the ventilation holes of the DAC.—Ed.)*

This device is highly recommended, as it delivers what it promises in a nice package at an outstanding price. The Link is a runaway bargain.

—***Glenn Strauss***

lenges of reception in a moving vehicle over an ever-changing terrain are so much greater. A cheap car radio needs to be "smarter," in some respects, than a $3K audiophile tuner!

If you have read David Rich's FM tutorial in Issue No. 23, you have some idea of what can go wrong in the IF section of an FM tuner. Blaupunkt has done an end run around the IF demons by digitizing the IF signal directly and performing all functions of FM demodulation and stereo decoding in the digital domain. This removes the distortion and noise associated with analog signal processing. A further benefit is that part of the IF filter can be folded into the digital domain, permitting very good selectivity and distortion characteristics. (The degree to which the IF stage can be moved to the digital side depends on the distortion and noise performance of the A/D converter.) I shall leave it to Messrs. Rich and Modafferi to explain and critique this technology; what I was interested in was how the Blaupunkt would perform when inserted in place of my accustomed audiophile tuners.

The answer is—better! I was able to receive more stations and cleaner signals than before. Note (a) that this may not have been due to the digital IF processing alone and (b) that the Blaupunkt needs a really fat signal before it goes into the stereo mode, thus giving the impression of lower noise on more stations. Even so, the few classical/jazz/talk local stations I listen to regularly were definitely received in stereo and came in cleaner than ever. There was velvet silence behind the signal and no crackly breakup for any reason at any time. That was a new experience. Switching on the fluorescent lights, which had always caused a momentary unpleasant sizzle with other tuners, resulted in a barely audible sigh. I was delighted.

As our longtime readers know, this kind of starry-eyed endorsement without detailed technical documentation is not our style. I may end up regretting it by the time our two RF experts are finished with the Blaupunkt. I have always been inclined to celebrate, however, when a few hundred dollars' worth of new technology leaves the multikilobuck stuff in the dust and I didn't want to hold my tongue until the next issue. If I turn out to have been wrong, I'll be the first to admit it. That's another thing our longtime readers know.

-Ed.

5-Channel Power Amplifier

Sherbourn 5/1500

Sherbourn Technologies, Inc., 6 Landing Lane, Hopedale, MA 01747. Voice: (508) 478-5296. Fax: (508) 478-5270. E-mail: sherbourn@aol.com. Web: www.sherbourn.com. Model 5/1500 5-channel power amplifier, $1750.00. Tested sample on loan from manufacturer.

Ron Fone, former CEO of Mcintosh who now heads up this fledgling amplifier company, would like to position the Model 5/1500 as a Bryston 9B ST challenger at less than half the price. I can neither dismiss that bit of audacity out of hand nor agree with it, really. The Sherbourn does resemble the Bryston in basic architecture; it even surpasses the 9B ST in sheer power; but the perfection of the latter is in the details and there the resemblance is, well, less than striking. Ron, whose affable Britannic diction resembles Michael Caine's, would undoubtedly argue over a pint of mild ale that the differences are unimportant and/or overpriced. A defensible point of view, I'll concede, but let the details speak for themselves.

The Model 5/1500 consists of five monoblocks, just like the Bryston it emulates. Each channel has its own power supply with toroidal transformer. Only unbalanced inputs are provided, thus lessening the appeal to the professional market, in contrast to the Bryston. A so-called dynamic clipping switch on each channel, when flicked on, allows the power supply to limit the current drawn by the input stage during hard clipping. (I never felt the need to use it.) With the cover off, the layout is reasonably neat and uncluttered, without quite equaling the sleek professional look of the Bryston. Taking the cover off is an ordeal because the butter-soft Phillips-head screws are torqued down like crazy in production and cannot be removed without the most brutal metal surgery. (When will hard, high-quality screws become an industry standard? It would only cost pennies.) The fit of the chassis metalwork is only so-so, but the overall build quality inside is quite nice. Basically, this is upscale consumer-electronics country, not the crossroads of the high-end/professional world.

The measurements I took were on the whole fairly impressive. Into 8Ω the distortion-versus-output curves bottom out in the -92.5 to -95 dB range at 165 W clipping; into 4Ω the minima range all the way from -87 dB to -93 dB at 250 W to 350 W clipping, depending on frequency. The exception, alas, is the 20 kHz distortion curve, which does not track the others but diverges from them even at milliwatt levels and bottoms out at —82.5 dB into both 8Ω and 4Ω Now -82.5 dB (0.0075%) distortion at 20 kHz is certainly not bad (who can hear the 40 kHz second harmonic, anyway?) but it is about 10 dB worse than the distortion at the lower frequencies, and that means dynamic distortion, which is generally a symptom of circuit-design shortcomings.

The PowerCube of the 5/1500 shows larger numbers—i.e., greater dynamic power—than that of the Bryston 9B ST (see above) but also identifies an obvious problem, which I am coming to in a moment. Into resistive (0°) loads of 8Ω/4Ω/2Ω/1Ω, dynamic power of the measured channel was 294W/512W/802W/1074W. That's a lot of short-burst power for the money. Into capacitive (-60°/-30°) and inductive (+30°/+60°) loads, dynamic power was slightly up at *8Ω* and *4Ω,* as it should be, but at *2Ω* and 1Ω the power inexplicably sagged when the load was *slightly* inductive (+30°) but not when it was *very* inductive (+60°). The problem is especially evident at 1Ω/+30°. I think there is a current-limiting glitch in the circuit design. No big deal but not as nice as it could easily be.

Crosstalk between any two channels

Do You Need a Spectrum Analyzer?

AudioControl Industrial SA-3052

Audio Control Industrial, 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. SA-3052 one-third octave real-time spectrum analyzer and SPL meter, from $1350.00 up (depending on options).

When I originally approached the Editor about a review of this product, he was reluctant. His primary concern was that the SA-3052 is not a laboratory-grade measurement instrument, certainly not when compared to an Audio Precision. That is an indisputable fact, but misses the point. The SA-3052 does not pretend to be a high-precision measurement device, but instead a powerful, complete, and portable tool for making in-room spectrum analyses of audio waveforms. Its accuracy is sufficient for that purpose, its features thorough, and its price within reach. Its portability and (optional) battery operation make it a complete package.

The SA-3052 is a 30-band -octave spectrum analyzer, SPL meter, and real-time store/retrieve display instrument. It is supplied with a condenser microphone using standard 12V phantom power. The microphone is accurate enough (±2 dB) over a range of about 40 Hz to 12 kHz or so, dropping off a bit only at the extremes. For other than nearfield precision measurement, this is plenty accurate, and more so than the ubiquitous Radio Shack SPL meter upon which countless audio decisions have been and are being based.

Spectrum measurements are made using the unit's Class 2 (IEEE) pink-noise sources to drive the device under test (DUT), or line-level outputs from the DUT to drive the spectrum analyzer. While the line-level measurement accuracy could help set up a PA system, it is useless to measure any professionally designed modern line-amplification device. I verified this by running the pink-noise generator into a Bryston BP25, and back into the SA-3052, which showed a 1 dB variation around 1 kHz; using my HP function generator into an IHF load, the Bryston is flat within my ability to measure accurately (0.1 dB) in the same range.

Where the SA-3052 excels is in obtaining rapid, easily repeatable sound *profiles.* By profile, I mean a general trend or sound impression at a given point in space. For setting up loudspeakers, this is a tremendously valuable tool. While some of the computer-driven tools David Rich examined in Issue No. 25 will help you locate your loudspeakers, the Audio Control will let you see how those placements pan out. (And keep in mind that the computer models do not work much above 300 Hz; those models will not assist in dealing with middle and high-frequency anomalies, such as slap echo.)

at 1 W output into 8Ω declines at the rate of 6 dB per octave as the frequency is lowered, typically starting with -65 dB at 20 kHz and ending with -103 dB at 20 Hz (average figures for a number of measurements). Nothing to be ashamed of there vis-a-vis the Bryston. Frequency response at 1 W into 8Ω is +0.0/-0.2 dB from 20 Hz to 20 kHz, -0.7 dB at 10 Hz, and -1.0 dB at 45 kHz. Nice and conservative.

In actual use I was very happy with the Sherbourn (get lost, tweaks, it has no sound of its own), except for one peculiarity. Often, but not always, the amplifier shuts down with a thump when the power is switched off—and the thump doesn't consistently come from the same channel! Relay problem? Maybe. This particular sample only? I wouldn't know. What I do know is that on the basis of big-clean-watts-per-channel-per-dollar the Model 5/1500 is very hard to beat. Slight warts and all, I can recommend it. I cannot and will not equate it, however, with the Bryston 9B ST.

—Peter Aczel

I was able to get rapid and outstanding results at the listening position by using the SA-3052's real-time and time-averaging capabilities. By placing the microphone at the listening chair, setting the display to "slow" mode, and making gross speaker movements, I could dial-in a good initial placement. I then made a number of measurements at microphone positions above, below, left, and right of the listening sweet spot with a vertical window of several feet, and a horizontal window of about six. Each spectrum analysis was stored in one of the six memories and retrieved later for review of the sound profile. After several profiling iterations, I obtained a position that represented the best compromise. This is a technique recommended to me by several recording engineers, and also practiced by John Atkinson at *Stereophile.* They all feel that it correlates with well-balanced transducer (microphone or speaker) placement.

The SA-3052 helped me identify and correct a vexing problem I had had for a while in one of my listening rooms with a very fine pair of speakers. Subjectively, the midbass sounded disproportionately left-channel heavy, which had the psychoacoustic effect of moving the whole image to the left. This was determined not to be caused by source imbalance, amp level errors, or speaker mismatch. But sinewave and handheld meter measurements proved too time-consuming, and I let the problem endure. With the SA-3052, I was able to see the problem's profile easily—a huge standing wave in the left side of the room. Shoving the speakers around changed the profile, but did not correct it. These particular speakers, with their powerful bass and dual rear ports, were not happy. But there was a happy ending—I got tremendous improvement by incorporating an easy-to-hide minisubwoofer, left wall, room-central. And what helped determine that this unusual spot was optimum? Yep, the SA-3052. Problem solved. Game over. Lights out.

I also liked the SPL measurement and display capabilities of this unit; press one button and you have a large digital SPL display visible anywhere in the room. Using a long XLR microphone cable (supplied), one can then position the mic at the listening spot and monitor the unit away from the speakers and the mic. This helps prevent midrange measurement errors introduced by body proximity that could result from a handheld SPL device. The unit also has peak-hold functionality.

While I have touched on the main functions likely to be of interest to readers of ***The Audio Critic,*** this unit can do many more things, including sound reinforcement measurements. It also has a standard printer interface option, optional VGA monitor drive, and a special model capable of measuring up to 175 dB! If the latter function causes you shivers of delight, you are obviously reading the wrong magazine.

While the price is high, so is the value. For the well-heeled audio enthusiast, this is a great accessory, and no toy. For the audio professional, such as a home-theater specialist, this is a must. One way to beat the price issue would be for a gaggle of audio buddies to split the cost and time-share it; unless you switch speakers frequently, not having it available on demand is not such a big deal. But to have it is to want it. Well done.

-Glenn Strauss

Outboard D/A Converter

Van Alstine Omega IV DAC

Audio by Van Alstine, Inc., 2202 River Hills Drive, Burnsville, MN 55337. Voice: (612) 890-3517. Fax: (612) 894-3675. E-mail: info@avahifi.com. Web: www.avahifi.com. Omega IV DAC outboard D/A converter, $999.00 (direct from Van Alstine). Tested sample borrowed from staff member.

The idea here was to do a quick reverse-engineering analysis of this thing to see if the company's extraordinary claims for the unit could be explained. The information I came up with appears below, with measurements made at ***The Audio Critics*** lab. No technical details were available directly from the manufacturer, and nothing can be learned from the company's Web site. That site contains nothing but gibberish from a technical point of view. Once we open the top of the DAC we see strange things.

Except for the standard Crystal CS8412 S/PDIF decoder (no 96 kHz sampling supported here) all ICs have heat sinks on them that obscure any identification. The CS8412 gets a single digital input from an RCA input jack. No Toslink input. No digital outputs for MD and CD-R equipment. Can you believe it at this price? Cheap twisted-pair cables connect the digital-in jack to the PC board. Coax cable of the correct impedance must be used if the eye pattern of the incoming signal is not to be degraded at the input of the S/PDIF decoder. (We are not talking audio here. Impedance match counts with high-speed S/PDIF signals.)

Two PC boards are observed with the cover off. In addition, a cheap single-secondary transformer with center tap is seen. One PC board is the power supply board, which is single-sided. It has on it a 6800 µF main filter cap for the positive supply rail, which is regulated down to 12 V by a 7812CT 3-terminal regulator. The negative rail uses a smaller 3300 µF main filter cap and a 7912CT regulator. Another 7805CT is on the main PC board and that's all, folks. Yes, the main filter caps are mismatched to save a couple bucks. The larger supply cap needs to supply the +12 V analog circuits as well as the +5 V digital supplies. The positive rail serves to keep the whole digital side alive and thus needs a bigger cap. I have never seen a power supply this cheap in any high-end unit I have ever examined. Measured results showed the 60 Hz and 180 Hz hum levels down 105 dB, which is just OK at best and far worse than we got from other cheaper units reviewed in this issue.

At this price point we expect, at a minimum, separate secondaries on the transformer for the analog and digital stage, and separate transformers are common. A ± 15 V analog supply is the expected norm, not the ±12 V used here. In regulators, ten are expected, not three. That means separate regulators for the DACs (dual mono), preferably double-regulated, as well as for the analog stages (dual mono), for the S/PDIF decoder, for the digital filter, and another for any glue logic. In this unit all digital logic, the S/PDIF decoder, the DAC, and the voltage reference for the DAC come off the same digital supply. It is an amazingly cheap design. What is particularly unbelievable is that the S/PDIF decoder and digital logic are on the same regulator as the DAC. The performance of the DAC will be affected by the dirty supply voltage, especially of the one Van Alstine has chosen, as we shall see in a moment. But before we get there let me report that the channel separation was OK at 102 dB in the midband but dropped to 66 dB at 20 kHz, which in part is related to board layout (see below) and in part to the power supply. A drop to 80 dB at 20 Hz indicates a coupling effect which is most likely power-supply related.

On the top side of the main board some ICs are socketed, some are not. Do not ask me why. No ICs should be socketed because it adds lead inductance and kills performance in high-speed mixed-signal designs. On the top side of the main board is a block of four identical gain blocks. Each uses an 8-pin DIP chip (socketed) and a 10-pin DIP (anybody want to guess what comes in a 10-pin DIP?) that is soldered directly to the board. No complex discrete circuits are seen anywhere, including these gain blocks. I did not spend any time to reverse-engineer the analog gain block. The S/PDIF decoder and a couple of TTL gates are also socketed. Four discrete transistors are seen. These turn out to be the solid-state switches for muting and de-emphasis. At this price we expect high-quality relays, not cheap transistors. That transistorized muting switch comes unglued when the plug is pulled, and the thing generates some very large oscillations that could take out a speaker if the volume was high when the power went off. This is a poor, cheap design.

Frequency response measurements show that the circuits do function. Things stayed within a ±0.1 dB range up to 10 kHz. At 20 kHz we are up +0.3 dB. De-emphasis response was similar. A 0.1 dB channel mismatch was worse than most and is likely the result of the DAC choice (see below).

Flipping the main board over exposes a 44-pin surface-mount quad flat pack. Also on the bottom side of the board are a lot of flying lead components that indicate problems on the original PC board, or redesign with a lack of funds to spin the PC board. The board is hand-soldered and the work is not of professional quality. The PC board is double-sided with through holes (good). These do not take kindly to hand soldering. The fact that the solder mask is missing from the board only makes things worse. That surface-mount IC should never be hand-soldered. One suspects that the chip in question only comes in a surface-mount package, since nobody doing hand soldering would ever elect to use a package that has such small leads and at such a fine pitch. Power-supply routing is also really strange, and lousy, for a high-frequency digital/mixed-signal design. With respect to what are obviously bypass capacitors, the board layout is poor. No electrolytic caps are seen on the main board except by the surface-mounted 44-pin chip. They should be near the gain blocks and the DAC but are not seen.

In modern designs, where wave soldering equipment is available, surface-mount bypass caps are used because they can be brought much closer to the active devices, thus reducing routing inductance. Van Alstine cannot do this because they are doing the soldering work in-house, by hand, in a very low-budget environment. They should be outsourcing the PC board layout and its assembly to a modern manufacturing site. These sites can run the low volumes Van Alstine is running, but the boards would cost more than what Van Alstine is paying for these do-it-yourself assembly jobs. At the $999 that Van Alstine gets for this unit, they should easily be able to cover the cost of a professionally laid-out and assembled PC board.

The way the 44-pin package is connected on the PC board identifies it as the digital filter. The pinout is identical to that of a Philips SAA7322, or SAA7323, dual filter/DAC. (We cannot tell by reverse engineering which is used, since they both have the same pinout.) As I suspected, the chip only comes in a QFP (quad flat pack) package. No other chip in the world would have this pinout. It has a 1-bit DAC on its silicon in addition to a filter, and it is an old design that is out of production. (With all the obsolete chips on this board, it looks like the main board design is at least 5 years old.)

They only hook up some of the components for the DAC part of the SAA7322, and this may be just to fool those doing reverse engineering. In reality the chip is only used as a digital filter. The choice of this digital filter is very strange. Philips designed the chip to be used as a filter *and* a DAC. They never recommended it just as a filter. I think Van Alstine wanted 4-times oversampling, not 8-times (who knows why, but they claim this is a feature on their Web site), and I think they needed a filter that did not require a microcontroller to drive it. Maybe they just think they like the sound of that filter, who knows? In any case, it does not support HDCD and its specifications are not in the same league with the top-of-the-line digital filters from NPC or Sony.

The DAC is identified by how it is connected up as the very low-end Philips TDA131X chip. The X stands for three different versions of the chip: 1, 2 or 3. I cannot tell by reverse engineering which is used by Van Alstine, since they all have the same pinout. All three are multibit units using MOS current copiers as the LSB element in the DAC core. Circuits of this type match only to the 12-bit level. The TDA131X chips were designed for use in portable equipment, where their small size (only 8 pins) and low power-supply operating voltages made them a star in 1992 when they were introduced. Now lots of 1-bit units work at this voltage and have this footprint. Philips specifies full-scale THD at the 10.5-bit level, which is the pits. It settles down to 13.5-bit levels as the digital input level is lowered, which is why the distortion is probably not audible—but why in the world anybody would use it in a high-end, or even low-end, powered product I do not know. In the data sheet for the TDA131X chips, Philips recommends 7 different digital filters for use with the TDA131X, and none of them is the old and discontinued SAA7322/7323 used by Van Alstine in this design.

And now—what does all this add up to? The worst measurements we have ever made. Gain linearity is off 0.4 dB at -70 dB, 2 dB at -80 dB, and 4 dB at -90 dB. And in case you think that was a defective channel, it was the good channel! The worse channel was off 0.4 dB at -60 dB, 1.6 dB at -70dB, 5.4 dB at -80 dB, and at -88 dB (just before -90 dB) we have a 21 dB (yes, 21 dB!) gain linearity error. Below -90dB the analog signal level stayed constant in both channels as the digital signal level was reduced. Perhaps a problem with the digital filter.

Looking at a 1 kHz signal at the -60 dB level in greater detail shows the 3 kHz 3rd-harmonic distortion component down by only 35 dB. Odd harmonics stay around 40 dB below the fundamental level out to the 17th—yes, 17th —harmonic. (OK, bring on the high-end folks who are experts on harmonic structure.) THD + N at the -60 dB level is essentially constant up to 10 kHz, with the worse channel hovering around 29 dB below the signal level. Pushing the level up to -20 dB gets out of the range of the DAC's awful linearity (remember it was designed this way for the reasons explained above); there we measured the THD +N to be 64 dB below the signal level in the less "good" channel. Again, it is essentially constant across the spectrum. Coming up to the 0 dB level (i.e., full scale) brings us face to face with the DAC again. The 2nd, 3rd, and 4th harmonics of a 1 kHz fundamental are down only 75 dB. Harmonics are visible all the way out to the 19th, which is at -91 dB. The 9th is at the same level as the 2nd. The flat level of the harmonics again points to linearity problems in the DAC. It all adds up to full-scale THD + N of—66 dB across the spectrum. No, that is not a typo: -66 dB THD + N for $999. (The Editor has been making faces for years when it's 20 dB better than that.) The THD is so high and so completely dominated by the DAC that other effects like clock jitter and dynamic distortion in the analog stage are not possible to observe. The THD just stays constant to 10 kHz.

Full-scale output level was the industry standard 2.0 V rms. It is nice to see that Van Alstine has chosen not to play with this to make the unit "sound different" (by modifying its output level a couple of dB). This is about the only nice thing that I can say, and one wonders why one should be nice at all. This is a company that claims that the Omega IV DAC is "easily the best piece of audio equipment we have ever designed, and likely the best at any price from any audio manufacturer." They say they "understand all significant non-linearities at all frequencies and amplitudes." They claim to have applied "thousands of hours of original advanced math circuit analysis programming." You can go to the Web site (www.avahifi.com) and see more. You are now armed with the truth about this unit. It is of questionable design and construction. Measured results confirm the analysis. Your choice as to what to do with respect to purchase of the Van Alstine Omega IV DAC should be clear.

—David Rich

We need a good review to sell this product. *We sure do.*

I'd like to see it reviewed by ***Audio.*** ***Audio*** *is out of business*

How about Fi? *Fi is out of business.*

Maybe we could send it to ***High Performance Review.*** ***High Performance Review*** *is out of business.*

Well, do you think ***Stereophile*** *would review it?* *Not expensive enough for them.*

Then how about ***The Sensible Sound?*** *Too expensive for them.*

OK, let's send it to ***The Audio Critic.*** *For heaven's sake, no! They'11 measure it!*

Peter and Tom Aczel

By PETER ACZEL, Editor

Direct Stream Digital and the "Super Audio CD"

Big-deal technology or bigtime marketing?

By now every audio publication has genuflected with varying degrees of piety before the Sony/Philips-sponsored DSD technology and its incarnation in the SACD format. Labels as diverse as Delos, dmp, Sony, and Telarc have adopted it; their renowned recording engineers have unequivocally endorsed it; at the same time more than a few technologists with the highest credentials (including Stanley Lipshitz, John Vanderkooy, and our own David Rich) have expressed serious reservations about it. The situation has been further complicated by the contradictory claims of superiority heard over and over again from the DVD-Audio camp, without any DVD-A players and discs available for comparison. (That ridiculous delay appears to be over just as this issue is going to press.)

The huge gap between this issue of ***The Audio Critic*** and the last (a problem now solved by our alliance with our new publisher) leaves us with an interesting perspective, having heard all the claims, read all the reviews, and talked to all the protagonists and antagonists. With all that accumulated information to check against our own lab-bench and listening tests of the flagship Sony player, I feel ready to draw some conclusions. I am not about to explain the technology of DSD and the SACD; it's been done. Sony's own magnificent 1999 brochure introducing the format and the SCD-1 player would take up most of our pages here if we just reprinted the text, without pictures. Ed Foster's exegesis, in conjunction with his Sony SCD-1 review in the November 1999 issue of the now defunct *Audio*, is also quite thorough and easy to understand, whether you agree with his value judgments or not. And there are others. I shall cut straight to the nitty-gritty.

I disagree with the initial assumption on which DSD is based, namely that 16-bit PCM at a sampling rate of 44.1 kHz leaves a great deal to be desired, even for simple 2-channel audio. The absolute best CDs recorded that way have been of unimpeachable fidelity, capable of yielding complete musical satisfaction. The trouble is that 44/16 leaves no margin for error, so that only the most careful and precise work will extract the potential quality inherent in the medium. Going to 20-bit digital processing makes the recording technique a lot more forgiving, allowing some tolerance in setting the 0 dB full-scale level and giving elbow room to 16 bits under all conditions. The best conventional CDs recorded at 20 bits (such as, for example, some of John Eargle's VR^2 recordings on the Delos label) make me question the burning need for a brand-new technology like DSD, certainly as far as transparency is concerned. What's more, the PCM approach can be extended to 24 bits and 96 kHz sampling, as exemplified by the Chesky "Recorded at 96/24" CDs and others, not to mention the 96/24 and 192/24 standards chosen for multichannel DVD-Audio. (There the difference is that the discs themselves can store 96/24 and 192/24 PCM signals, unlike the 44/16 CDs originating from 96/24 digital tapes.) In short, multibit PCM is alive and well and not a technological problem. That goes for any number of channels up to 6 or possibly even more.

So, why did Sony and Philips come up with the idea of encoding music as a noise-shaped 1-bit signal at a sampling rate of 64 x 44.1 kHz = 2.8224 MHz and eliminating digital filters? After all, there is nothing really revolutionary about either 1-bit quantization or oversampling. The two electronics giants have many technical answers, some more convincing than others, but the real answer has nothing to do with engineering and everything to do with business. Copy protection is considerably easier with DSD than with multibit PCM, and intellectual property is a bigger business issue than sonic fidelity. Breathing new life into the Sony/Philips CD format as the original patents expire is possibly another business issue.

What are the basic features Sony and Philips cite in favor of the SACD format? One is audio-perfectionist purity through simplicity—look Ma, no intermediate conversion stages, no decimation filter, no interpolation filter. Another is bandwidth (potentially 100 kHz, tradable against dynamic range). Still another is greater resemblance to analog signal processing, requiring only a lowpass filter at the output to restore the analog signal. (Hey, *digital* is still a bad word in some circles.) What are the main objections of the skeptics? Essentially the same as they always have been to delta-sigma modulators in general and 1-bit delta-sigma in particular. One recurrent complaint is that the delta-sigma noise floor is not white but tonal (idle channel tones, limit cycle oscillations) and particularly so with 1-bit conversion. A major criticism specific to DSD is that the high-frequency noise power rises very rapidly just above the audio band because of the very high-order delta-sigma modulator, so that drastic lowpass filtering is needed to protect the downstream electronics and transducers—in which case where is the claimed bandwidth/phase advantage? (See also below.)

A complete analysis of the alleged shortcomings of the Sony/Philips technology is beyond the scope of this discussion—too technical, too many pages—and would furthermore elicit serious, and far from incompetent, disagreements from the DSD advocates. I happen to be impressed by the credentials of the objectors—they also include other important names I have not been authorized to print—and must further note that the industry has been moving away from the 1-bit solution, 3 bits and 4 bits being the current preference for delta-sigma. On the other hand, how flawed can DSD be when John Eargle, Tom Jung, Michael Bishop, and other recording engineers of their caliber are happily using it and praising the sound to the skies? Interesting question, isn't it, considering that digital editing of DSD without conversion to PCM is not even possible at the present time.

I'll leave it at that and proceed to my test report.

Sony SCD-1

Sony Electronics, Inc. 1 Sony Drive, Park Ridge, NJ 07656. Voice: (201) 930-1000. Fax: (201) 358-4060. Web: www.sony.com. SCD-1 Super Audio CD Player, $5000.00. Tested sample on loan from manufacturer.

Frankly, I'm surprised this $5K fantasy product is still in the line. The SCD-777ES, originally introduced at $3500, is virtually the identical player except for the cosmetic differences, and its price keeps coming down. I have seen it advertised on the Internet for $2700. That's powerful intramural competition. I must admit, however, that the SCD-1 is absolutely the sexiest piece of audio equipment I have ever laid eyes on. The fabulous brushed metal finish, the flush control buttons all on top, the huge display window, and above all that massive sliding panel, like the door of a bank vault, slowly revealing the play mechanism—it all made me think of Q when he says, "Now pay attention, double oh seven." If I were half my age and single, I would perhaps consider the $5000 a good investment to impress young lady visitors.

A big letdown, on the other hand, even before any performance evaluation, is the stereo-only design of the SCD-1. The Super Audio CD was conceived and extolled as a multichannel medium—where are the other channels? Coming, we are told. (In the immortal words of Henry Miller, any more than Christmas is coming?)

The default configuration of the SCD-1 has a 50 kHz lowpass filter at the output to keep out the high-frequency noise discussed above. Sony warns the user not to disable this protective device unless the amplifier circuits and the tweeter have been designed to handle high power at very high frequencies. Such special designs being the exception in nearly all foreseeable installations, I obediently left the filter in place for my measurements and listening tests. Sony provided a Super Audio CD test signal disc, as well as a number of hybrid dual-layer SACDs on which both the DSD and standard CD versions of the same music were accessible. My own battery of standard test CDs and a number of regular music CDs completed the available software.

I was somewhat underwhelmed by the SACD measurements. The frequency response surpassed conventional CD players with a —1 dB reading at 35 kHz and only -0.1 dB at 20 kHz, but no other performance characteristic did. Full-scale THD + N hovered around -85 dB at all frequencies, and the FFT spectrum of a 1 kHz tone at 0 dBFS showed 2nd, 3rd, and 5th harmonic components all around —95 dB (provin thar's HD in them thar N). Furthermore, this wasn't gain-related analog distortion because reducing the digital input to -20 dB and then to —40 dB effected no improvement. Quantization noise measured -87.6 dB (which is consistent), the unweighted dynamic range 94.4 dB. Even with that 50 kHz lowpass filter in place the out-of-band noise appeared to be some 30 dB above the in-band noise level.

Crosstalk ranged from -87 dB at 50 kHz to -126 dB at 10 Hz, declining in an almost straight line (very nice but no better than many). And, yes, I almost forgot: gain linearity error is ±0.0 dB at all levels, which is perfect, but then it is inevitably very good with 1-bit quantization, whether the equipment comes from Sony or Joe Blow.

The conventional CD measurements, on the other hand, are pretty close to state-of-the-art. Full-scale THD + N stays very close to -95 dB across the audio spectrum. Unweighted dynamic range is 96.7 dB; quantization noise measures 96 dB. The FFT spectrum of a dithered 1 kHz tone at —60 dB ("Rob Watts test") shows a bin-by-bin noise floor of -124 dB and no harmonic blips whatsoever. Gain linearity error is again nonexistent at any level (a 1-bit DAC remains a 1-bit DAC).

What I was most interested in, of course, was whether or not the DSD version of the same music sounded better than the PCM version. The availability of both versions on the dual-layer SACD was no help. You can't just toggle between SACD and CD with the control buttons; you have to stop the player and start from scratch in each playback mode. That takes more than half a minute, and by then your auditory memory cries uncle. Ken Pohlmann, in his September 1999 *Sound & Vision* review, pretty much gave up on this problem. If I believed in the conspiracy theory of audio (like some 'philes I know), I'd suspect Sony of not wanting me to make an objective comparison.

What I had to do was to obtain a few standard CD releases of the same recordings and then synchronize and level-match my reference CD player with the SCD-1 for quick A/B switching. It wasn't a double-blind listening test but at least it was a valid A/B comparison. What did I hear? Absolutely no difference. Both A and B sounded gorgeous in each and every case. Every once in a while I thought the SACD had a bit more air, delicacy, and definition way up there in the highs, but then I listened some more and decided it was just wishful thinking. In all fairness, I must add that my high-frequency cutoff is undeniably lower than it was when I was twenty-five. If and when it appears that the SACD is here to stay—meaning second- and third-generation players and lots of new releases—I plan to organize a series of ABX sessions with seasoned (but not ancient) listeners.

Until then, let's wait for the arrival of DVD-Audio and see which format sticks to the wall. Both? Most unlikely. Neither? I wouldn't be the least bit surprised. TAC

BOX 978

(continued from page 3)

The Audio Critic:

I had barely started reading the latest of your all-too-infrequent issues [No. 25], when I spied my name in a letter to the Editor. Following the text, I found that something I had written had been misconstrued. I have always prided myself on what I thought was clear, straightforward writing, so here I hope to set the record straight.

The issue, in the John Ötvös letter (p. 5), is timbre matching of the front and side channels, and he refers to an article of mine in *Audio* (May 1997). There, while discussing the THX embellishments to Dolby ProLogic, I take issue with the need for electronic timbre matching, by equalization. I point out that "Sounds arriving from the sides, or even from random incidences, cannot and should not match the timbre of sounds arriving from the front. It (timbre matching) is not natural—the complex shape of the external ears ensures that." To avoid ambiguity, my parenthetic addition should have been in the original.

Timbre is a perceived attribute. If *identical* sounds from *identical* loudspeakers should arrive at the ears from the front, and subsequently from the sides, they will be perceived to have different timbres. The reason is that the sounds are modified in different ways as they diffract around the head and ears on the way into the ear canals, and to the eardrums. This is absolutely natural, and the different timbral signatures, among other things, allow us to correctly localize the directions from which the sounds came. This is something that we should not, and need not, meddle with.

To prove this for yourself, play some music with lots of high frequencies or, better yet, some pink noise, through a single loudspeaker. Start by facing it, as if it were a center channel. Then rotate the head until the loudspeaker is off to the side. Listen to the change in timbre, or spectral balance. It is not subtle, but it is absolutely natural, and not something that needs to be "fixed."

Along the way you can perhaps hear that the timbre has changed perceptibly at angles much less than 90 degrees. The fact is that identical left, center, and right loudspeakers, because of their differing angles of incidence to the ears, do not generate *absolutely* identical timbres. We accommodate all of this without alarm, because it is all part of real life.

If anyone should wish to equalize sounds arriving from different angles in ways that make them sound alike, this is an artistic judgment, and such equalization should be incorporated in the recording itself, not imposed on the playback system as a permanent distortion of reality.

So, just to make the point absolutely clear, all loudspeakers in a multichannel system should have the same (neutral) spectral behavior, whatever their directional characteristics. When those sounds arrive at the ears, coming from different directions, they will not have identical perceived timbres. But, there is nothing to "fix" with equalization, because there is nothing wrong. All is as nature intended.

Sincerely,
Floyd E. Toole
Vice President Engineering
Harman International Industries, Inc.
Senior Vice President
Acoustical Engineering
Harman Consumer Group

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

High-Tech Gear for Your TV Room (a.k.a. Home Theater)

DVD VIDEO PLAYER

Denon DVD-5000

Denon Electronics, a division of Denon Corporation (USA), 19 Chapin Road, Pine Brook, NJ 07058-9777. Voice: (973) 396-0810. Fax: (973) 396-7448. Web: www.del.denon.com. DVD-5000 DVD Video Player, $2500.00. Tested sample on loan from manufacturer.

This was intended to be Denon's "statement" DVD player for 1999-2000; unfortunately it has been discontinued, probably because the market appeared to be too limited for such a high-priced unit. It still makes sense, however, to review it briefly as an example of Denon's current engineering approach, which you can expect to find, perhaps with some economies, in the latest generation of Denon players.

The DVD-5000 gives the appearance of deluxe equipment right out of the shipping carton. The sleek black chassis weighs over 36 pounds, definitely dreadnought class. The front panel proudly sports the DVD, CD, Dolby Digital, DTS, THX Ultra, and HDCD logos; the back panel reveals digital inputs as well as outputs, so you can use the DVD-5000 as an outboard D/A converter if you wish. The front-panel display isn't overly complicated, as in some players, yet it is clear and informative (it even indicates 96 kHz sampling and 24-bit quantization in linear PCM sound, when present). The available screen menus are rather elaborate, offering great flexibility of setup and formatting; confusion is minimal, however, because the displays are logical and basically self-explanatory. The remote control is similarly ergonomic and not intimidating to the new user.

David Rich's once-over of the circuit design is in the accompanying sidebar.

Circuitry of the Denon DVD-5000

We have not spent a significant amount of time to make a value judgment on the video circuits. The operation of the video channels is similar to that of the audio channels. There are DACs, which in this unit are all in a single Motorola chip. The claimed 10-bit DACs are really 9-bit, and the data sheet gives little information on dynamic performance. Separate DACs provide C, Y, Pr, and Pb. These are then used to develop the various video outputs (C and Y for S-video; Pr, Pb, and Y for component video). A variety of discrete and integrated amplifiers and buffers lie between the DACs and the jacks. Like the audio sections, the video should be characterized in terms of SNR, frequency response (for video amplitude and phase), and linearity performance.

On the audio side things are very nice. The digital filter is the Pacific Microsonics PMD-100 that will decode HDCD discs. This chip is used in conjunction with a Denon proprietary DXP6001AF, which does the company's Alpha processing. Burr-Brown PCM1704J DACs are used, two per channel; µPC4570C's do the I/V conversion; Analog Devices OP-275 op-amps do the differential-to single-ended conversion. A GIC filter is built with the two op-amp sections of an NE5532. Another OP-275 buffers the output and is coupled to the output with back-to-back electrolytics bypassed with a smaller film capacitor. A relay does the muting function. All the analog circuitry in the audio path is powered from its own transformer. For some reason analog rails are only ±12 V. The DACs need ±5 V and they get it from subregulators that are in a dual mono configuration.

You may ask about the other three channels. The answer is there are no other channels. Denon's theory is that DVD videos should be decoded in the AV receiver or preamp. Two-channel audio is better converted in the DVD player and then, using the analog pass-through mode (if it exists), the signal goes to the MDACs and then on to the two power amps. Given what we have seen in typical AV products this makes sense. Nothing like the analog stage of this unit is found in those products. ***—David Rich***

I played Joe Kane's "Video Essentials" on the DVD-5000 just to see if there were any video anomalies that could be directly attributed to the player; I could not find any. As a CD player, the Denon's measurements equaled, but did not surpass, the results I had obtained with the best players of other makes. Full-scale THD + N averaged —95 dB at frequencies below 1 kHz (about 3 dB short of the theoretical ideal). Above 1 kHz the distortion started to climb, reaching a worst-case level of-85 dB at 10 kHz. This, however, proved to be gain-related analog distortion because at —20 dBFS it disappeared altogether; at that level THD + N was the same for all frequencies. Quantization noise was in the —96 to —97 dB range; dynamic range measured 98 dB (can't ask for better). The mono tonicity waveform was just about the best I have ever seen. The error-correction torture test on the "CD-Check" disc of Digital Recordings was passed by the Denon on Levels 1,2, and 3; Level 4 caused an occasional small click; Level 5 clicked like a castanet. (CD players that play Level 5 without a click are supposed to exist but I have never tested one.) In the outboard D/A converter mode (for which I see very limited use, if any) the DVD-5000 yielded excellent measurements consistent with the foregoing, and that's all that needs to be reported under the circumstances.

If Denon's latest DVD players (and inevitably coming DVD-Audio players) turn out to be of comparable quality, we can all be very happy.

—Peter Aczel

Personal TV System

ReplayTV 2020

ReplayTV, Inc., 1945 Charleston Road, Mountain View, CA 94043. Voice: (650) 210-1000. Fax: (650) 691-0094. Web: www.replaytv.com. 2020 personal TV system, now superseded by the similar 3020, $499.99. Tested sample owned by reviewer.

In Issue No. 25 we looked at the Sony SLV-M20HF VCR with the Gemstar Guide Plus+ program guide. What we have here is the next step. Two big problems are addressed by ReplayTV, which is a startup company. One is the limited guide size in the Gemstar system. It is missing channels and only has only a two-day time span. This limited guide size is caused by the limited bandwidth of the data transmission used by Gemstar. The ReplayTV unit calls during the night to a central number and downloads two weeks' worth of guides for all channels in a program area. This idea clearly adds to infrastructure costs compared with Gemstar. These added costs are supposed to be paid for as part of the up-front cost of the system. Some advertisement mechanism is also said to be possible, but I have not seen it yet. The competitive TiVo system has a monthly fee, and if that does not turn you off, the TiVo system reports viewing information back to the central site. Big Brother has arrived at TiVo. Multiple ReplayTV systems can be used in a house without a collision to use the phone, it is claimed, but I got only one unit to play with so I cannot verify this.

The second problem addressed is the problem of the taped-based VCR itself. If we accept that a VCR is mostly a time-shifting mechanism—and this becomes especially true with a strong program guide—then we realize that tape is the wrong medium. A single tape does not hold enough hours of programming, and navigation is lousy, with slow access time. More problems come from insufficient information about the recordings and where they are on the tape, even with the improvements Sony tried to make with their SmartFile on the VCR we tested last time. So ReplayTV dispenses with the tape and gives us a big hard drive. Now we are in the land of digital video, since a hard drive will not do analog recording. So video signals get digitized, compressed into the MPEG-2 format, and written to the drive. Output from the drive is also decompressed and converted back to analog. It is amazing that this technology is available at this price. The majors saw digital recording technology becoming available to consumers years from now—and they saw it tape-based. Digital VCRs from the big boys have been shown in prototype form for a few years at the CES, with all sorts of hints at copyright problems and four-figure prices. The Far East crowd has now been broadsided by the American startups. Panasonic and others are now licensed for ReplayTV technology and use the ReplayTV infrastructure and program guide. Sony is licensed for the awful TiVo, which is another US-based company, although one with close ties to the entertainment industry (which helps explain many of its undesirable properties).

The hard drive of the model reviewed here holds 20 hours of video, although a newer 60-hour model has meanwhile also become available. That is a huge amount of data, which can only be stored economically as a result of the recent advances in hard-drive technology that have also brought down hard-drive costs. Unfortunately, although it is a huge amount of data, 20 hours turns out to be merely OK. More would be better, since using up 20 hours is a lot easier than one would think. What one winds up doing is selecting more and more shows for recording, since this unit uses the infallible one-touch operation (like the Sony SLV-M20HF) from the remote. Touch the remote twice and your shows get recorded week after week. Now, you do not find yourself watching all the shows—sometimes just parts, like that one segment of *60 Minutes*—but you do wind up with a lot of stuff on the disk. The problem gets worse as you delay playing back and then deleting a show, waiting for a rainy day in the future.

The navigation guide is very well done, blowing away not only the Gemstar guide but also the guide in the General Instruments DCT 2000 digital TV cable box. (I have no experience with satellite systems.) Shows are clearly described, with the detailed description of a selected show placed well on the screen. The search system worked well, although the letter selection using the four-key cursor control is not a fun way to spell out words.

The space on the ReplayTV unit is too limited to save something permanently as one would do with a VCR. The unit does have outputs to drive a VCR for permanent storage. At slow tape speed settings, the results are less good than direct to tape from the live signal. The limited bandwidth of the VCR, its noise and signal distortion apparently make the invisible artifacts of the MPEG coder visible. Running the VCR at faster speed solves the problem.

Three recording modes that trade density against quality are available (the 20-hour time limit is for the highest density mode). The quality of the picture coming off the unit in its high-density mode looks fine to me, as good as the cable signal I have coming in. (Mr. Bill, can you say low signal-to-noise ratios?) I saw no new digital artifacts (I am using a DTV with a line doubler). I have no doubt that with a better incoming signal some degradation might be apparent (why would they include the low-density modes if this was not the case?), but things are much better than a VCR running at standard speed.

If you have a set-top box or satellite box, you will need an IR blaster supplied by ReplayTV. An IR blaster is a fiber-optic cable with an end that is glued over the IR receiver of the box to be controlled. An IR transmitter in the ReplayTV unit drives the fiber-optic cable. In this way the ReplayTV box acts like a human at the remote control, selecting the correct channel to be viewed or recorded.

The ReplayTV unit can also do an interesting trick for real-time viewing. If you use the unit as a tuner for watching TV (which implies you are looking at the picture from the encode/decode chain and its artifacts, if any), then you can pause the show you are watching and search backward in time at speeds up to 20x. This allows for methods to handle interruptions and do instant replays. Obviously, tricks like this require the hard drive to read and write at the same time. The head actually can do this at digital video data rates. The hard drives are modified for Replay audio uses. They have reduced rpm's compared to PCs, for lower heat and noise.

If you pause the unit for a few minutes before the start of a program, you can push the commercial skip button, which advances time by 30 seconds. Obviously, this feature also works when replaying a recorded program. TiVo lacks this feature; indeed, TiVo is claimed to be able to record commercials under software control and replace live commercials with these taped commercials, chosen on the basis of your viewing habits as monitored by TiVo. (You want this thing in your home? It is not coming to mine!) TiVo has many other operational differences, including the fact that it never spins down the hard drive, which reduces its lifetime; furthermore, it has a live TV buffer of only 30 minutes. There are also many features missing in TiVo, including the missing one-touch recording mode that is the essence of these devices.

Channel surfers will find the ReplayTV's slow response to a channel change (you have to wait for the MPEG chain to settle down) maddening. Of course, you can view the on-screen guide to see what is on and not, instead of surfing. Fast searching through a program goes up to 16x, which is far faster than a VCR. The unit has slow-motion modes as well. It all works a lot better, both in terms of the video on the screen and in terms of initiating the operations, than any VCR I have used.

Build quality is that of a computer, not a piece of consumer electronics. The PC board is double-sided. The chassis is a thick metal unit. The construction is USA. Think high end. I am not going to give a detailed circuit description because I do not have a schematic. Given the number of pending patents and trade secrets imbedded in this startup product, I did not even think to ask.

Problems include the high noise level from the hard drive. When powered off, the hard drive starts up a couple times an hour. Navigation through the on-screen guide is complicated by slow response time to remote commands. Online customer service is good, with short wait times and a competent, really helpful staff. Unfortunately, they have not been able to resolve some channel assignment problems at the upper end of the band. Thus, in another proof of Murphy's Law, the Ovation network is unusable through the box. (This is the only all-arts cable channel. Lots of stuff from the U.K. that PBS did not pick up for some reason.) I also found that the box may not be demodulating the upper bands correctly. Dialing channel 97 brought in channel 100. The telephone dialer does not have a pulse mode, thus requiring that you have tone dialing.

Overall, ReplayTV will change the way you relate to television. You watch when you want to watch and you do not have to watch the commercials. Interruptions that used to send you scrabbling for a VCR tape are no longer a problem. No doubt this technology will find its way into satellite receivers and set-top boxes, eliminating the need to have yet another piece of equipment in the media room, but I would not recommend that you wait. At this price ($499.99 for the new 3020 version), it is time to make a place for it on the top of your set now.

—David Rich

DVD Video Player

Toshiba SD-5109

Toshiba America Consumer Products, Inc., 82 Totowa Road, Wayne, NJ 07470-3191. Voice: (973) 628-8000. Fax: (973) 628-0372. Web: www.toshiba.com. SD-5109 DVD video player, $999.99. Tested sample on loan from manufacturer.

To put all the positive things about this piece of equipment into one sentence, it is a progressive-scan DVD player, which is still the exception rather than the rule, and it performs very well on DVDs and creditably on CDs, at a far from low but nevertheless reasonable price. That's a mouthful but it's basically the whole story.

The SD-5109 will decode Dolby Digital, DTS, and HDCD, but it is not a THX-certified player. Its video outputs include composite video, S-video, and interlaced as well as progressive component video. Its audio outputs comprise Bitstream/PCM coaxial and optical, 2-channel, and 5.1-channel surround. Some kind of preamplifier is still needed because the 2-channel and 5.1-channel outputs aren't volume-controlled, so you might as well connect one of the digital outputs to a full-fledged home-theater processor with more than just entry-level decoding capabilities. A special feature of the SD-5109 is its dual disc system—you can load two discs so you won't waste your valuable double-feature viewing time getting up from the couch and swapping discs. The mechanical construction of the player appears to be on the flimsy side; it weighs less than 8 pounds, not even one fourth the mass of the high-end Denon reviewed above. Its low mass has no apparent effect on its performance, however.

My evaluation of the video performance of the SD-5109 was inextricably linked to that of the Toshiba TW40X81 rear-projection TV (see below). The progressive-scan component-video outputs of the DVD player were connected to one of the component-video inputs of the TV, and Joe Kane's "Video Essentials" test disc and a number of high-quality DVD movies were played. Without a completely equipped video laboratory (there may be one in our future) it was impossible to determine to what extent the observed performance was due to the DVD player or to the TV. As reported below, the overall video performance was about as good as it gets with consumer-grade equipment. If there had been a problem with the DVD player, it would have shown up in the combined results.

As a CD player the Toshiba did not set any records in measurable performance, not even for players in its own price category, but the results were still quite respectable. The least impressive measurement was full-scale THD + N. At no frequency was it better than -85 dB and at 10 kHz it rose to -71 dB (27 dB excess distortion). Nor was this just gain-related analog distortion, as is usually the case, because the -20 dBFS measurement, while only 4.5 to 6.5 dB short of the theoretical ideal across most of the audio spectrum, still showed 8.5 dB excess distortion at 10 kHz. Consistent with this was the quantization noise: —85 dB. Dynamic range, normalized from the -60 dB level, measured 97.7 dB (fine). Gain linearity, generally expected to be perfect with the multibit delta-sigma DACs used in the Toshiba, was indeed so: -0.15 dB error at -90 dB in one channel, 0.0 dB in the other. The monotonicity waveform showed some departures from perfection but nothing shocking. As for error correction, Levels 1, 2, and 3 on the "CD-Check" disc of Digital Recordings were negotiated without a click; Level 4 caused some clicks; Level 5 was unplayable, with constant clicks. That seems to be par for the course (i.e., good).

Bottom line: the Toshiba SD-5109 may not be the most impressive DVD/CD player out there, nor even the best value per dollar, but it does the job—a very complex high-tech job—and it has progressive scan, which may be more important to you in the long run than any other criterion.

—Peter Aczel

Rear Projection TV

Toshiba TW40X81

Toshiba America Consumer Products, Inc., 82 Totowa Road, Wayne, NJ 07470-3191. Voice: (973) 628-8000. Fax: (973) 628-0372. Web: www.toshiba.com. TW40X81 16:9 ColorStream rear projection TV, $2800.00. Tested sample on loan from manufacturer.

We're like Republicans and Democrats, unlikely to agree. I'm talking about those of us who favor direct-view TV, the bigger the better, and those who must have the really *huge* screens and therefore opt for rear projection with its undeniable drawbacks. (Front projection is not an issue—we'd all love to have it if we could afford it and had a dedicated media room with a 66-by-122-inch Stewart screen.) I happen to be an independent and have criss-crossed repeatedly from one camp to the other; right now I have an elderly 40-inch direct-view heavyweight in my home theater system.

I found the 40-inch wide-screen rear-projection Toshiba somewhat difficult to evaluate because it is both bigger and smaller than my TV and produces a better and less good picture. Huh? What I mean is (a) that in the 4:3 "Standard" mode the picture area is smaller and in the 16:9 "Theater Wide" mode it is bigger and (b) that the on-board line doubler of the Toshiba provides higher, more lifelike resolution but the picture is still not as bright and crisp and comfortably viewable. If I were a truly intense techno-video geek, I would rate the TW40X81 higher—but with the lights on, beer and snacks on the coffee table, and people walking in and out, I'd rather watch the Super Bowl on the big direct-view tube.

The line doubler of the TW40X81 is strictly entry-level, not claimed to be in the same league with even the least pretentious Faroudja or suchlike, but it works very well just because the screen is relatively small. I haven't seen the 65-inch and 56-inch models in the same Toshiba series, but I suspect that the identical line doubler makes their pictures just a little coarser. The other advanced feature of the TW40X81 is "ColorStream," which is just a promotional moniker for component video. There are two sets of HDTV-ready component video jacks alongside the usual S-video and composite video inputs. The component video inputs will sense 480i (interlaced) or 480p (progressive) scanning in the input signal and engage or disengage the line doubler accordingly. High-definition 1080i input signals (from an external source) remain in that format, but 720p is rescaled to 1080i (as in most HDTV-ready sets). I used the progressive outputs of the **Toshiba SD-5109** player (see above) to view DVDs via the component video inputs of the TW40X81; for the VCR I used one of the S-video inputs. At this point in the era of diversified video signal sources the "cable-ready" TV is an anachronism, so the everything-ready Toshiba only tunes up to channel 13. You, the owner, must provide the cable converter box, satellite receiver, digital video receiver, set-top box, or whatever. I had to use both available 75Ω antenna inputs to switch between the internal tuner and my cable box. Going from, say, channel 6 to channel 34 requires fairly intensive menu maneuvering.

That brings me to one of my major misgivings about this and all other similarly configured new TVs. You have to be at least slightly geeky to be comfortable with the computer-like user interface. In my home theater system I have five program sources; some people have more. Their outputs go to different input modes of the TW40X81, which in turn has three selectable picture sizes, one of which has three selectable modes. Just to adjust the set for the program material and the correct aspect ratio with proper image proportions requires serious navigating from menu to submenu. And that's just the beginning because there are also other choices to be made. To optimize one given viewing situation to the total satisfaction of a video perfectionist goes into menu ramifications that cry out for a macro—but the set has no storage capability except for channel programming. Don't buy a TW40X81 for your grandparents on their golden anniversary; I don't think they'll be able to figure it out. I barely did.

Just because the screen is wide and there are all those picture-size option, it doesn't follow that every program can be displayed with the exact aspect ratio and exact image proportions of the original release. Even with vigorous menu massaging you may end up with something slightly cut off or slightly stretched or both. Sometimes it's best to split the difference. Wide-screen TV is lots of fun—I'll even say, hard to be without once you've had it—but not perfect. As for the picture quality itself, it's damn good with the set right out of the shipping carton and can be further tweaked with "Video Essentials," Joe Kane's marvelous video optimization DVD. Some videoniks hold the Socratic belief that the unoptimized picture is not worth viewing. I refuse to be quite that dogmatic—this being an audio magazine that acknowledges the ascendance of home theater, not a publication for AV techies—but even I must concede that the factory default setting of contrast is for maximum brightness, not the finest detail, probably to be at least feebly competitive with direct-view sets. Also, none of the three default gray-scale settings appears to yield truly correct color temperature at various contrast levels, but we have no color analyzer in our laboratory (not yet, anyway), so I did not get involved in sophisticated gray-scale massaging. I found no significant geometrical distortions and did not find it necessary to readjust convergence. After doing the best optimization I could with the tools at hand and viewing some high-quality DVDs, I came to the conclusion that I had never seen a better rear-projection picture—perhaps for no other reason than a relatively small screen in combination with 7-inch tubes—but also that I am still not a rear-projection enthusiast.

Yes, I just said this is an audio magazine, so I should say something in conclusion about the Toshiba's built-in pair of 5-inch speakers, driven by 14-watt amplifiers. As they say in South Philly, fuhgeddaboudit!

—***Peter Aczel***

LAST-MINUTE NEWS:

Just before press time, John Ötvös, president of Waveform, made the shocking announcement that the company would be closed down. As our readers know, the Waveform Mach 17 has been our reference speaker since 1997. For details of this sad development, and of an inventory closeout sale, go to www.waveform.ca.

By Tom Nousaine

It Never Gets Better

I've heard it all when it comes to refutation of controlled listening tests. The reason why no one has shown a single example where wire, amplifiers, or bits have any audible consequence not related directly to readily measurable elements is—listener anxiety . . . small sample sizes . . . large sample sizes . . . slow switching . . . fast switching . . . switch connections . . . experimenter bias . . . poor program material . . . well, you get the idea. Some people, mostly those who make and sell high-end gear, or publish magazines catering to that market and their customers, refuse to accept any data (even their own), no matter how carefully gathered, that doesn't support their preconceived notions.

On the other hand, practically any anecdote that *does* is accepted as confirming data. No one ever accuses a believer of being full of beans about audibility, no matter how ridiculous the claim. Occasionally there will be a teapot argument about why a given inanimate object has audible effect, but statements about audibility are never taken to task.

This fits in with recent analysis of junk science (cold fusion, perpetual motion machines, alien visitation, et al.), where the evidence never gets any better. Spacecraft sitings have been reported for decades, but so far no one has ever produced a convincing photograph or artifact that we have ever had one visit from space. The evidence is always just around the corner, enticingly close, soon to be revealed—but, in fact, it never gets any better.

The audibility-without-known-cause case is exactly the same. When someone conducts a controlled experiment with null results, the experimenter is accused of being sloppy, having a bias that colors the results, or practicing "bad science." These complaints would sound more reasonable *if* there were convincing contrary evidence. But so far no one has produced a single replicable experiment where audibility without reference to level, frequency response, or operating error has been confirmed. Although this is a dead horse, I bring it up as a prelude to the new millennium where things are definitely getting better.

How so? Well multichannel audio has a big foothold. Just as moving from mono to stereo improved things, even more channels are the basic way to increased sonic realism.

Powered speakers are digging in. Incorporating the electronics into the design of the speaker offers a large performance improvement potential. In the '30s the modern moving-coil loudspeaker was arguably the highest-fidelity component in the audio chain. Since then, most development has been concentrated on storage and transmission media, so the loudspeaker is now orders of magnitude behind everything else, except microphones. Electronic control is the best tool for near-term performance enhancement of speakers, and you'll see more and more of it.

Although it may take a generation to work out the details, a revolution in media access and distribution is taking place. People now download music with ease, accepting a small reduction in quality for low-cost access. Even quality has a new dimension. With analog storage media, such as LP, there was a 3-10% distortion level always present. With data-reduced digital media the program may be perfect 90-97% of the time. This is a whole new concept of distortion, free of the tyranny of the storage medium itself.

We'll also see a move to per-use access of programs. Rentals have always been a good choice for video-based programs. Pay-per-view has been shown to be a great alternative to owning, for sporting events and movies. The Circuit City disaster was just the beginning of the process of nonownership access to programs. Revolutions often fail the first time.

Quite frankly, I don't feel the need to own programs anymore. If I had instant access, I'd actually rather *not* own them. I think we'll see a transition to per-use access as up/download capacity improves.

The other thing I think will happen is a move toward integrated audio/video systems. People purchase computer systems from a single source as opposed to a bunch of pieces integrated at the consumer site. Didn't people always purchase audio systems from a single store? Yes, they did, but most people think of the PSB speakers matched to the Denon electronics as separate things. I always do.

The home-theater-in-a-box with electronics (and sometimes program source) built in is just the start of the audio/video system as an appliance trend. Indeed, I've now come to view my playback equipment as appliances. Functionality and convenience become bigger factors as performance develops to commodity levels. People will eventually buy their entertainment systems as a single integrated and branded unit.

Sonic performance has become a commodity with well-designed modern products. Consumer amplifiers are commodities these days. A well-designed one sounds just like all the rest. They vary in power output, of course, and we buy them on that basis, but as they become part of an integrated system who will know or care?

We will also see more rapid introduction of new formats. Technology cycles halve with every new generation. For example, the golden age of LP lasted about 30 years, from about the mid-'50s to the mid-'80s, when CD replaced it. Same for cassette, the primary analog consumer tape format. It took roughly 30 years from inception until it was replaced by CD-R.

The replacement for CD is DVD, which arrived in 1998, about 15 years after CD started. Expect the next major media in 2004-6. SACD? Two-channel is dead. DVD-A? Maybe, but the de facto standard is Dolby Digital and it's not slowing anybody down. What we are seeing, however, is a trend toward more formats more quickly developed. Eventually a single processor will just decode anything thrown at it. When we buy access (as opposed to owning a chunk of plastic), the decoding instructions will come with the software and we won't care anymore at the consumer end.

So what should we make of all this as we get on with our audio lives? Life is great. Things are getting better performancewise. All our existing stuff will work in the future as well as it ever did. Today even our mono is the best it's ever been.

However, expect the bullshit quotient to increase exponentially with progress. When equipment reaches commodity levels in performance, it can only be sold with promotion. To informed enthusiasts there's nothing wrong with that. Readers of *The Audio Critic* have the best bullshit detectors on the face of the planet. Let's just keep them in good tune.

Capsule CD Reviews

By Peter Aczel, Editor

Because of the long interval between the last issue of ***The Audio Critic*** and this one—not to recur again under our new publishing regime—some of the CDs reviewed below are very recent releases and some are older (but still not ancient history). Note that the year in parentheses after the CD number is the year of recording, not the year of release.

Cedille

Cedille Records is the trademark of The Chicago Classical Recording Foundation and a label new to our pages (perhaps out of sheer negligence).

Franz Liszt: *"Liszt for Two." Six Hungarian Rhapsodies for Piano Four-Hands; Mephisto Waltz for Two Pianos. Georgia & Louise Mangos, duo-pianists. CDR 90000 052 (1999).*

These are not transcriptions by some hack arranger. Like Bach, Liszt transcribed his own compositions frequently. All of this music exists in versions for solo piano, piano duo, and orchestra, all from the hand of the composer. Twenty fingers create more of a wow effect than ten possibly can, and the Mangos sisters specialize in Liszt *and* wow. These performances will knock your socks off. I don't how good the two Chicago ladies are in more "spiritual" music, but in these showoff pieces they are simply diabolical. A few hundred years ago they would have been burned as witches. The recording also helps; it demonstrates how good the Schoeps MK2 microphone can be on piano music—the attack transients and dynamics are awesome, the bass is thunderous. What more can I say? The Beethoven Op. 111 it ain't, but boy, is it fun!

Chesky Records

This label prides itself on cutting-edge technology. They have switched entirely to 96/24 recording.

David Chesky: *The Agnostic. Slovak Philharmonic Orchestra and Slovak Philharmonic Choir, Stephen Somary, conductor. CD202 (2000).*

David Chesky cannot be accused of modesty. In his eclectic, basically nondissonant, but still contemporary-sounding style he has composed a gigantic work (and I mean Havergal-Brian-gigantic) for orchestra, chorus, and soloists, with libretto by (yes!) himself. Such a Wagnerian ego invites either adulation or ridicule, depending on the stature of the man's work, so I am not about to stick my neck out critiquing *The Agnostic* just a few months after its debut. I'll just say that it held my attention on first hearing. The recording is, appropriately, extremely spacious, with a huge dynamic range and clean climaxes. I wish it were a little drier and more intimate, but that may not have been possible with that many performers in a big reverberant hall. The orchestral playing and the singing are good enough to embarrass my innate Hungarian condescension to all things Slovak. As for having your own compositions recorded on your own label, Wagner had Bayreuth built for his own operas, so there.

The Cleveland Orchestra

This is not a CD label, but the ten-disc set of previously released Decca recordings below is now available only from The Cleveland Orchestra.

"The Cleveland Sound," *a 10-CD box set.* ***Anton Bruckner:*** *Symphonies No. 3, 4, 5, 7, 8, 9.* ***Gustav Mahler:*** *Symphonies No. 1, 4, 5, 6. The Cleveland Orchestra, Christoph von Dohnányi, conductor. 466348-2 (1988-94).*

There may be a small number of better performances on CD of each of these ten symphonies, but there is no better orchestra than the Cleveland. Their burnished tone, breathtakingly virtuosic precision, responsive teamwork, huge unstrained climaxes, and just plain musicianship are unsurpassed, perhaps even unequaled. Dohnányi may not be the world's most exciting conductor but he never falls below a very high level of technical competence and he is the perfect curator of the Cleveland sound. Even if these recordings are the only ones you own of these vast orchestral works, you will not be shortchanged and you will certainly know how they should sound—and sound is a very important part of their impact. The recordings are in the Decca multimiked idiom, with which it is possible to disagree, but of their own kind they are as good as it gets. This would make a princely gift to the Bruckner/Mahler lover.

Decca

This renowned label used to be called London in the United States but after the PolyGram shakeup it reverted to its native appellation.

Béla Bartók: *The 6 String Quartets. Takács Quartet: Edward Dusinherre, violin I; Károly Schranz, violin II; Roger Tapping, viola; András Fejér, cello. 289 455 297-2 (2 CDs, 1996).*

This formidable music, arguably the most important composed for string quartet since Beethoven (hey, I have authoritative support for my Hungarian cultural chauvinism), no longer sounds so formidable. The now only half Hungarian Takács ensemble plays it with such technical aplomb and lovely tone that the fierce "modernity" of the string writing begins to come off as familiar mainstream. For someone who cut his teeth on the classic 1972 Juilliard and stunning 1988 Emerson recordings this is not a replacement but a worthwhile addition, especially since the sound quality is more up-to-date—vivid, close-up, but without any unpleasant harshness in the loudest passages.

Gioachino Rossini: *II Turco in Italia. Cecilia Bartoli, Donna Fiorilla; Alessandro Corbelli, Don Geronio; Michele Pertusi, Selim; Ramón Vargas, Don Narciso. Orchestra and Chorus of the Teatro alia Scala di Milano, Riccardo Chailly, conductor. 289 458 924-2 (2 CDs, 1997).*

This is an absolute delight. The opera itself should be much better known because it effervesces like Rossini's very best and breaks into luscious song every few minutes—but it's the singing that makes this performance special. Bartoli is merely the greatest Mozart/ Rossini mezzo of our lifetime—she just has to open her mouth and you are spellbound—and the rest of the cast is almost as good. Chailly and the orchestra also sound as if they were having more fun than salaried employees should be allowed, and the recording is completely free from the slight zinginess that sometimes mars Decca's otherwise excellent sound. When everything is perfect, all you can say is *bravi!*

Delos

John Eargle still has my vote as the king of recording engineers, even if he likes to experiment with not yet mature technologies, such as DSD. A great cook knows what the dish is supposed to taste like, regardless of the skillets and kitchen utensils employed, and a great recording engineer knows what the recording is supposed to sound like through the loudspeakers, regardless of the recording hardware. John knows.

Gustav Mahler: *Symphony No. 2 in C Minor ("Resurrection"). Heidi Grant Murphy, soprano; Petra Lang, mezzo-soprano; Dallas Symphony Chorus, David R. Davidson, director; Dallas Symphony Orchestra, Andrew Litton, conductor. DE 3237(2 CDs, 1998).*
Symphony No. 3 in D Minor. Nathalie Stutzmann, contralto; women of the Dallas Symphony Chorus, David R. Davidson, director; Texas Boys Choir, ferry Bierschenk, director; Dallas Symphony

Orchestra, Andrew Litton, conductor. DE3248 (2 CDs, 1998).

Mahler is *the* classical composer for the hi-fi era, no question about it, and Litton has simply too much competition in these works. He is convincing enough; he deserves credit; he even has some inspired moments; but he is not quite a world-class Mahlerian. On the other hand, the engineering is world-class here, offering some of the best orchestral sound I have heard on CD. John Eargle has refined his technique in this recording venue (McDermott Hall at the Meyerson Center), and the results are simply magnificent. The balance between close-up instrumental detail and hall ambience is perfect. Incidentally, I happen to prefer these earlier symphonies to the neurotic, ranting and raving late ones.

Dmitri Shostakovich: *Chamber Symphony, Op. 110a.* **Alfred Schnittke:** *Concerto for Piano and Strings. Moscow Chamber Orchestra, Constantine Orbelian, pianist & conductor. DE 3259 (2000).*

For the first time, Delos uses the highly regarded Skywalker Sound studio of Lucasfilms, said to be ideal for recording smaller ensembles. This is stated to be a DSD recording, although it is not clear to me whether or not there was also a conventional PCM system running at the same time. The Shostakovich work is actually the composer's Eighth Quartet transcribed for string orchestra with his approval. It is a very serious work, not at all tovarich-friendly like, say, his Seventh Symphony. It makes a strong musical statement. The Schnittke concerto is not in the same league, at least in my opinion—but then Schnittke still knows more about music than I do. Orbelian is Delos's new golden boy, an American of Russian-Armenian extraction, now transplanted to Moscow. He undoubtedly has star quality, and his orchestra plays with great passion and precision. The slashing string attacks are captured by John Eargle with stunning fidelity, without any harshness even at peak levels. That's the special audio appeal of the disc, but the musical appeal is far from negligible.

Dmitri Shostakovich: *Festive Overture, Op. 96; Concerto No. 2 in F Major for Piano and Orchestra, Op. 102; Symphony No. 5 in D Minor, Op. 47. Dallas Symphony Orchestra, Andrew Litton, conductor/pianist. DE 3246 (1998).*

Litton is a Shostakovich specialist and here he presents the composer, very effectively, in three totally different moods: grandly ceremonial (the overture), brilliantly playful (the concerto), and dead serious (the symphony). That the Dallas orchestra plays well isn't news, but Litton's fleet-fingered, stylish piano playing is (at least to me). The performance of the centerpiece of the CD, the popular Fifth Symphony, is as good as you are likely to hear anywhere today—maybe not on the Mravinsky level, but he is dead. The recording is recent Eargle/Dallas, meaning the best there is (see Mahler above).

Nippon Columbia's label has few, if any, international superstars on its current roster, but the second-team players they do have sometimes rise to very impressive heights. Fame is not an infallible critic.

Claude Debussy: *"The Complete Solo Piano Works" (continued). Suite bergamasque; Deux arabesques; Danse bohémienne; Ballade; Reverie; Valse romantique; Nocturne; Mazurka; Danse (Tarantelle styrienne); Pour le piano. Michel Béroff, piano. CO-18047 (1995-96).*

This is very distinguished Debussy playing. Béroff has the big-time technique and the understanding of the Debussy idiom to give us very complete realizations of this unique music. He makes even the most familiar pieces sound fresh, spontaneous, and newly illuminated. Stupendous finger work and light pedaling are part of his secret. On top of everything else, the recorded piano sound is state-of-the-art.

Richard Strauss: *Symphonic Poems- II: Ein Heldenleben, Op. 40; Don Juan, Op. 20. Orchestre de la Suisse Romande, Eliahu Inbal, conductor. CO-18083 (1995).*

In Issue No. 25 I treated Volume I of Inbal's traversal of the Strauss tone poems *(Also sprach Zarathustra, Till Eulenspiegel)* somewhat unenthusiastically Volume II comes from the same week or two of recording sessions in Geneva, but I warmed to these performances much more readily. Orchestras and conductors have their days, even concert halls do (temperature, humidity, etc.); whatever the reason, I feel these are world-class performances by a maestro I have always admired, and the audio quality is also outstanding (e.g., the battle scene in *Heldenleben).* Inbal is always insightful, sensitive, and never cheap or obvious; if his orchestra were on the level of Chicago or Cleveland or Berlin, this CD would be right up there with the best in my book. Not that the Suisse Romande isn't a good orchestra, but good is not great, and a Richard Strauss score needs great.

This great classical music label also changed hands in the PolyGram shakeup; it is now part of Universal Classics.

Béla Bartók: *Bluebeard's Castle, Sz. 48 (Op. 11). Jessye Norman, Judith; László Polgár, Bluebeard; Nicholas Simon, Prologue. Chicago Symphony Orchestra, Pierre Boulez, conductor. 289 447 040-2 (1993).*

A dream performance? It certainly has some of the ingredients of one. Nobody conducts twentieth-century music better than Boulez. No orchestra plays better than the Chicago. No soprano sings better than Jessye Norman (at least when that big voice is warmed up). Only a few other recordings in the original Hungarian include the spoken Prologue. So why do I still have some reservations about this splendid production? As one of the 4½ or so Hungarian-speaking critics who can possibly be expected to comment on this CD, I have a problem with Jessye Norman's pronunciation. Mind you, she makes a brilliant effort to sound like a native and in more than a few phrases she almost does. She must have had some excellent coaching. But the rest of the time she is unable to wrap her tongue around the unfamiliar non-Indo-European words and the effect is comical to the Hungarian ear. This is not just faultfinding by a crotchety linguistic pedant. The Magyar cadences of the superb Béla Balázs libretto—Hans Christian Andersen-like in its fairy-tale magic—form the basis of Bartók's vocal metrics. The Hungarian vowels are in effect part of the music. (Imagine *La donna è mobile* sung with a Japanese accent and you'll begin to understand what I mean.) Other than that, I must call this the most magnificent performance of Bartók's early masterpiece I have ever heard. Jessye Norman's singing (qua singing) is simply wonderful, László Polgár is the best Bluebeard since the great Mihály Székely; the Chicago orchestra and Boulez are an incredible powerhouse; and DGG's "4D Audio Recording" captures it all with impeccable fidelity. Ignore my quibbles and get this CD.

A Tom Jung recording on this label is a guarantee of up-front, ultra-high-definition, demo-quality sound. The music is usually not my cup of tea, but there are exceptions from time to time. He also happens to be one of the early adopters of DSD.

"Far More Drums." *Eight pieces for percussion instruments. Robert Hohner Percussion Ensemble, Robert Hohner, director. CD-527 (2000).*

You want drums in your room? Tom Jung will put drums in your room. Boom-boom drums, tap-tap drums, subtly scraped thingamajigs, all as real as if they were next to your elbow. The Hohner players are as virtuosic as it gets. This is a 6-channel DSD recording, mixed down to stereo. I wonder about the editing equipment. David Kawakami, the Sony DSD guru, and Ed Meitner, the converter guru, are listed in the credits.

"Sacred Feast." *Gaudeamus choir, directed by Paul Halley. CD-526 (1998).*

Tom Jung recording sacred music sung a cappella in a chapel? Now I've heard everything. Yes, it sounds glorious, with just the right natural reverb, as if Tom had been making classical recordings in churches for years. The music is by composers as varied as Bruckner, Duruflé, and Messiaen; the Gaudeamus ensemble consists of 30 voices and is a beautifully disciplined, superb-sounding choir. Indeed, if you asked me to name a classier, showier choral recording, I wouldn't know where to begin.

Dorian

Craig Dory is extremely proud of his 24-bit digital recording technology called xCD, a "giant leap" according to the Dorian blurbs. I have always thought that the giant leap was Craig's original recording technique back in the late '80s, as it was unquestionably far ahead of the standard practice of those days. I discern only smaller incremental leaps since then—not that I have a problem with that. The following Dorian releases are in the xCD series and sound just great to me, regardless of leap size.

***"Danzón:** Music of Latin American Masters. " Simón Bolivar Symphony Orchestra of Venezuela, Keri-Lynn Wilson, conductor. xCD-90254 (1997).*

So this is what a symphony orchestra sounds like in xCD. Stunning, I must admit, with incredibly dynamic, unstrained climaxes. The sound is a little drier, more closely miked than Craig Dory's early recordings—and then I see in the credits that he didn't even participate in the session; Douglas Brown was in charge. This became one of my handful of orchestral demo discs after just one listen. The music is by seven different 20th-century Latin American eclectics, all of it most craftsmanly, foot-tappingly enjoyable, and not very profound.

***"Ocean Grove:** French Spectaculars on the Great Ocean Grove Auditorium Organ." Gordon Turk, organist. xCD-90267 (1998).*

A 9000-pipe big-mother organ originally designed about a hundred years ago is not so great for Bach but just what the doctor ordered for Widor, Vierne, & company, as in this program. The playing is highly competent and the recording by Craig Dory awesome, especially the 32-foot pedal stops, but I wouldn't trade you one of my favorite little Baroque organs against five of these monsters with their electric-buzzer-like tone envelopes.

***Antonio Vivaldi:** Concerti for Strings. Les Violons du Roy, Bernard Labadie, artistic director. xCD-90255 (1998).*

Why is Vivaldi so popular? Because he gives you the Bach texture without the Bach structure, which is intellectually demanding and not for everybody. In other words, Bach Lite. (Just a theory of mine, no disrespect intended.) Les Violons du Roy are Dorian's house ensemble, fifteen Canadian musicians who play with invariably lovely tone, considerable virtuosity, and secure musicianship. Here they play nine Vivaldi compositions featuring different string combinations, all of them delightful. The xCD recording by Craig Dory in a Québec church is extremely vivid, full-blooded, and up close, very different from what he does—or used to do—in the Troy (NY) hall. I can't imagine a more lifelike, believable string sound—another instant demo CD.

Harmonia Mundi

I am in total awe of this label. In the world of classical recordings, they are the equivalent of a five-star-rated restaurant. In culinary terms, their philosophy as I see it is: only the best materials (music), the best preparation (performance), the best service (recording). They even have the best promotional literature. Here are two of their *grandes spécialités de la maison.*

***J. S. Bach:** Saint Matthew Passion. Ian Bostridge, Evangelist; Franz-Josef Selig, Jesus; Sibylla Rubens, soprano; Andreas Scholl, alto (countertenor); Werner Güra, tenor; Dietrich Henschel, bass. Chorus and Orchestra of the Collegium Vocale of Ghent, Philippe Herreweghe, conductor. HMC951676.78 (1998).*

If a better performance of this masterpiece exists, I am unaware of it. Herreweghe delivers everything needed in this music—authentic style, devotional dignity, drama without lapses of taste, fine choral work, excellent instrumental support, beautifully played obbligatos. The solo singers are uniformly good; a countertenor in the alto arias is a bit unusual but with the great Andreas Scholl it works. The recording is transparent and utterly natural in sonic texture. A great addition is a marvelous interactive CD-ROM (for PC/Windows only) that tells you more than you'll ever want to know about Bach, about the background of the Passion, about the text, about the musical structure of the work, about Herreweghe's approach—shall I go on? There's nothing else like it, except... (see below).

***W.A. Mozart:** Così fan tutte. Véronique Gens, Fiordiligi; Bernarda Fink, Dorabella; Werner Güra, Ferrando; Marcel Boone, Guglielmo; Pietro Spagnoli, Don Alfonso; Graciela Oddone, Despina. Kölner Kammerchor, Concerto Köln, René Jacobs, conductor. HMC 951663.65 (1998).*

I would never have imagined that an even better performance of *Così* than Mackerras's on Telarc would make its appearance on CD only five years later, but this is it, at least to my ear. The period-instrument approach, with which I usually have some problems, works to perfection here under Jacobs's baton. I use the P-word advisedly because the man is a perfectionist—every hair is exactly in place, instrumentally, vocally, stylistically, in the use of ornamentation and the pianoforte, the whole bit. From the opening bars of the overture I marveled at precise attacks and releases of the virtuoso chamber orchestra. The singers are extremely fine, and Véronique Gens maybe a little better than extremely fine. The sound, as recorded in the studio of the Cologne radio, leaves nothing to be desired in immediacy, definition, and transparency. This is the Mozart opera that, in some highly respectable opinions, goes a step beyond the others in sheer musical inspiration, and it is a rare delight to hear it produced on this level excellence. And that's not all. An interactive CD-ROM, this time for both the Windows and Macintosh operating systems, accompanies the set. You can explore the opera scene by scene, number by number, follow every word with the bilingual libretto, get involved in the background of the work, enjoy a full-featured Mozart biography, and more. What a package! What music!

I have until now neglected this 20-year old English classical label, distributed by Harmonia Mundi USA. I can offer no justifiable reason and intend to mend my ways. The following is a great sample of what they do.

***Franz** Schubert: Piano Sonata in A Minor, D784; Piano Sonata in B-flat Major, D960; Piano Sonata in C Major, D613. Stephen Hough, piano. CDA67027 (1997-98).*

Stephen Hough is an intellectual, a scholar, a musician's musician, and above all a wonderfully sensitive pianist. The immortal B-flat Major sonata takes up well over half of this CD, and offhand I can't remember a better performance of it. There is an utterly secure, unmannered, natural flow to Hough's playing, allowing Schubert to emerge while the marionette strings of the interpreter remain invisible. That's not easy; only a few (such as Schnabel) could ever do it. I am seldom moved to tears, but this does it for me. The earlier sonatas are played just a beautifully, and the somewhat mellow (i.e., not overly clangorous) piano sound captured by the engineers suits the playing perfectly. The average level of the recording is a little lower than usual, requiring a higher setting of the volume control.

Mapleshade

This label is Pierre Sprey's domain. He is the man who starts with live-to-2-track analog and ends up with digital, viz. the CD. Don't ask me why, don't ask me how, just ask me if the end result is any good. Yes, very.

***Harold Ashby:** "Just for You. " Harold Ashby, tenor sax; John Hicks, piano; Keter Betts, bass; Jimmy Cobb, drums. 06232 (1998).*

Nearly all the great tenor saxophonists are dead. The young ones aren't great. There is still Harold Ashby, white-haired and well into his 70s, left over from Duke Ellington's great band and still playing in the grand tradition. Seven of the ten compositions on this disc are his own, one is the Duke's ("Sultry Serenade"). I like best his slow, bluesy opening number, "Reminiscing," and the Billy Strayhorn ballad, "Lotus Blossom." So, real jazz is still being played, maybe not at the pinnacle of the art, but all is not lost. The recording is right up there with Pierre Sprey's best—great presence, authentic tenor-sax timbre, stunning drums.

Naxos

Klaus Heymann's marketing lesson to the ailing classical CD industry has not been heeded by the major full-priced labels. Their sales are soft and Naxos is now Wal-Marting them, so to speak, out of a number of important international markets. We are the beneficiaries because Naxos quality remains high, in terms of both music and engineering, and the growing volume allows them to maintain the delightfully low price.

Samuel Barber: *Orchestral Works, Volume 1. Symphony No. 1, Op. 9; Symphony No. 2, Op. 19; The School for Scandal Overture, Op. 5; First Essay for Orchestra, Op. 12. Royal Scottish National Orchestra, Marin Alsop, conductor. 8.559024 (1998).*

This is the young Samuel Barber of 1933-47 (he died in 1981). The music was good enough for Toscanini and it's good enough for me—eclectic/romantic, beautifully orchestrated, easy to enjoy. The Scottish orchestra is absolutely first-rate; the 20-bit recording in a good hall by the prestigious Tony Faulkner is of demo quality. No bargain-basement production, this one.

Edward Elgar, Anthony Payne: *Symphony No. 3. Bournemouth Symphony Orchestra, Paul Daniel, conductor. 8.554719 (1999).*

Elgar's last symphony existed only in the form of fragmentary sketches when he died in 1934. This is in effect a symphony by Anthony Payne, based on what he believes the Elgar sketches might have turned into had they been elaborated by the composer. In other words, there is a lot more Payne in this Elgar than there is, say, Süssmayr in die Mozart Requiem. The net result sounds quite grand and Elgarian, if occasionally a little boring, to my not particularly Elgar-attuned ears. I must leave it to the specialists to critique this effort. The excellent orchestra appears to be more than equal to the task, and the carefully balanced Tony Faulkner recording is also equal to the best standards of today.

RCA Victor

In today's shrinking classical music market RCA Victor has a tremendous advantage. Instead of spending big bucks on a new recording with expensive superstars and high-cost paraphernalia, they can always dip into their bottomless reserves of classic performances and resurrect one with the aid of new technology, dehissing, digital remastering, etc. There isn't a music lover out there who has heard them all.

Anton Bruckner: *Symphony No. 4 in E-flat Major ("Romantic"). Berlin Philharmonic, Günter Wand, conductor. 09026 68839 2 (1998).*

Bruckner, according to the famous musicologist Alfred Einstein, "produced his most harmonious work in his Fourth Symphony, which depends almost entirely on beauty of sound." This recording possesses beauty of sound to the nth degree. The Berlin Philharmonic is unsurpassed in its string and brass sonorities, and the state-of-the-art BMG recording captures both the huge dynamics and the exquisite nuances. The bass line is particularly rich. But the real hero of this recording is the 86-year old (at the time of recording) conductor. I firmly believe that if the music world had received its first exposure to Bruckner from Günter Wand, the composer would never, never have been accused of incoherence in his gigantic symphonic structures. Wand knows the exact tempo and inflection, from bar to bar and phrase to phrase, to make the music sound all of one piece. It's magic. I cannot imagine any Brucknerite not rating this performance right at the top of the heap in a highly competitive field. Don't deny yourself this experience.

"Caruso 2000: *The Digital Recordings." Arias from operas by* ***Verdi, Puccini, Leoncavallo,*** *et al. Enrico Caruso, tenor; Vienna Radio Symphony Orchestra, Gottfried Rabl, conductor. 74321-69766-2 (1906-1920 and 1999).*

You simply must try this on an unsuspecting opera lover. You start Track 1 and a good orchestra plays the introduction to *La donna è mobile* in excellent 1999 digital sound. Then the tenor comes in—God, what a voice! But wait a minute, couldn't the engineers have done a bit more justice to that fabulous singer? What kind of crummy microphone did they use for voice—it rolls off and has a somewhat hollow, constricted coloration. Get a Neumann, turkey. After a minute or two, of course, the listener gets wise. The voice part of the recording is 1908 Caruso minus all the hiss, crackles, and pops; the orchestra is 1999 Vienna. Digital magic has been able to lift that glorious voice out of the early discs' background crud and mix it with a separate recording of the orchestra. Can you imagine the difficulties of a present-day orchestra trying to follow the dead Caruso's far from metronomic beat, not to mention the technical problems of finding the right stylus for nonstandard grooves, dealing with unpredictable deviations from 78 rpm, adjusting to old tuning pitches, etc., etc.? Digital technology and computer software apparently conquered all. The net result is that, for the first time, I am able to listen to a Caruso recording without wanting to leave the room. I can almost forget the slight acoustical disconnect between voice and orchestra. I actually find this CD thoroughly enjoyable. Furthermore, I can now say with a fair degree of certainty that of all the legendary tenors—Caruso, McCormack, Gigli, Björling, and successors—Caruso had the greatest throat, i.e., the most naturally beautiful and effortless voice, but was far from the finest musician. His distortions of the music as written wouldn't be tolerated today. Still, you've got to hear this one.

Igor Stravinsky: *Pétrouchka (original version); Le Chant du rossignol; Feu d'artifice. Vienna Philharmonic, Lorin Maazel, conductor. 74321-57127-2 (1998).*

You hardly ever hear *Petrushka* in its 1911 original version. Too many extra orchestral players, too expensive. Stravinsky himself revised and simplified the score in 1947, primarily for crass commercial reasons. This performance is the real McCoy, the dazzling work that straddled the dividing line between traditional and modern music. Some still think it is Stravinsky's masterpiece; at the very least it is a stupendous orchestral showpiece. You don't think of the Vienna Philharmonic as a Stravinsky orchestra, and that's one of the good things about this recording. They don't take the music for granted; every bar is played as if it were new and fresh to them (of course it isn't, not quite). That makes it a wonderful performance. The playing is both affectionate and virtuosic, and Maazel holds it together beautifully. The same can be said of the other performances, making this an early-Stravinsky feast of the highest order. The *Chant,* especially, has some fabulous melodic lines, colors, and sonorities. The recording is by the same engineer as the Bruckner I praised above but a little brighter in sound, as befits the orchestration. It is as clean, detailed, and powerful as the absolute best of any other label you can name. Another RCA Victor/BMG winner.

Reference Recordings

From audiophile cult label to almost mainstream—that has been the impressive progress of Tarn Henderson's and Keith Johnson's RR company. I say mainstream because they are now doing the big symphonic staples; I say almost because they are into HDCD and that sort of thing, unlike the majors. "Professor" Johnson has extremely high standards of recorded sound, so you are most unlikely to go wrong with an RR recording, techie/tweako frills and all.

Anton Bruckner: *Symphony No. 9 in D Minor. The Minnesota Orchestra, Stanislaw Skrowaczewski, conductor. RR-81CD (1996).*

Bruckner composed the same symphony nine times and finally got it right—so goes the waggish musicological commentary. No, I don't agree; but yes, this is probably his greatest. My problem is that Günter Wand (see above) has spoiled everybody else's Bruckner for me. This is actually a very fine performance, beautifully phrased and intelligently proportioned, by an excellent orchestra. Still, the flow of the music under Skrowaczewski's baton has a stop-and-go quality, typically Brucknerian, that Wand knows how to even out, subtly and naturally. It seems that the Austrian symphonist requires conductorial schnitzel power rather than kielbasa power (Phil Niekro's T-shirt slogan) for best results. The recorded sound is nothing short of superb, even without HDCD decoding.

Gustav Mahler: *Das Lied von der Erde. Michelle DeYoung, mezzo-soprano; Jon Villars, tenor; The Minnesota Orchestra, Eiji Oue, conductor. RR-88CD (1999).*

Mahler must be coming out of the ears, nose, and elbows of every reviewer these days; still, I make it a point to check out every new recording of *Das Lied* because it is a wonderful piece of music, difficult to perform on the level of excellence it deserves. It is also the exception to my frequently expressed reservations about late Mahler; its intensity is utterly sincere and convincing—it doth not protest too much, methinks, and the dark side is balanced by moments of delicate loveliness. Villars does not screech and scream like so many tenors in this work, but the high tessitura is not really to his liking; he still sounds uncomfortable in many passages. DeYoung sings quite beautifully, perhaps with a bit too much tremolo here and there; in the *Abschied* she gives the music everything she's got. Much of the credit must go to Oue, who shapes the work with great sensitivity and gets first-rate playing out of the various orchestral choirs. As for the recorded sound, even without HDCD decoding it blows away any other tecording of *Das Lied* I can think of; the richness of the upper bass and lower midrange is especially remarkable. Orchestra Hall in Minneapolis seems to have excellent acoustics.

Telarc

This was the leading-edge audiophile label of the early CD era, offering much better sound than the entrenched majors. The world has meanwhile caught up with them, without surpassing them. Their engineers are still in the forefront but no longer alone. As for worthwhile music to record, it's getting tougher every year, for them and everybody else.

***Ludwig van Beethoven:** Symphony No. 5 in C Minor, Op. 67; Symphony No. 7 in A Major, Op.92. Philharmonia Orchestra, Benjamin Zander, conductor. CD-80471 (1998).*

Who needs another recording of these symphonies? Those who think they have heard them the way Beethoven wrote them. No other music has been interpreted and reinterpreted as much as these classics, to the point where some performances are unrelated to what modern scholarship can clearly prove were Beethoven's intentions. Benjamin Zander likes to be a teacher to the music-loving public (not unlike the late Leonard Bernstein) and here he undertakes the task of teaching us what the real Beethoven, without the accumulated layers of questionable performance "tradition," sounds like. It isn't the same as period practice (à la Norrington or Gardiner)—the world-class Philharmonia Orchestra is all modern instruments—it is more a reexamination of lost details and correct tempi. Zander devotes an entire bonus CD in this set to a discussion of his research, insights, and goals, so there is no need for me to say more. The main differences from the listener's point of view are highly transparent, almost x-rayed orchestral textures and generally, but not invariably, faster-paced movements than expected. These are extremely strong, intelligent, effective performances without being on the Carlos Kleiber level of overwhelming artistry and impact. The recording by Tony Faulkner in a very fine London hall is the cleanest, most dynamic, most naturally musical I have heard in a Beethoven recording. You need to check out this one.

***Antonín Dvorák:** Stabat Mater. Christine Goerke, soprano; Marietta Simpson, mezzo-soprano; Stanford Olsen, tenor; Nathan Berg, bass-baritone; Atlanta Symphony Orchestra and Chorus, Robert Shaw, conductor. 2CD-80506 (1998).*

Dvorák in a tragic vein? He tries but it doesn't come to him naturally, even though he composed this music at a time of multiple personal tragedies. It is a massive and somber work, but from time to time the fundamentally upbeat nature of the composer shines through. He couldn't help it; he lacked the required self-pity later institutionalized by Mahler. Although the music never quite rises to the heights of the Cello Concerto and the best of the symphonies, it is quite beautiful in many passages; the writing for the quartet of solo singers is particularly fine and extremely well sung in this performance. The recently deceased Robert Shaw has been the gold standard in choral work for more than half a century, so *that* part of the recording is an absolute given. I have always found a slight suggestion of boxy thickness in Telarc's choral recordings in Atlanta whenever the lower registers get loud, and this is no exception—even though the recording is DSD, converted to the CD format via Sony's Super Bit Mapping Direct, no less. Other than that it is simply a well-engineered contemporary choral recording without extraordinary sonic qualities, at least to my ears.

Water Lily Acoustics

Kavichandran Alexander draws on the ancient wisdom and aesthetics of his native India and the wizardly technology of the wicked West to make the presumably purest of purist recordings on this label.

***Franz Liszt:** Les Préludes. **Antonín Dvorák:** Three Concert Overtures—In Nature's Realm, Op. 91; Carnival, Op. 92; Othello, Op. 93. The Philadelphia Orchestra, Wolfgang Saw allisch, conductor. WLA-WS-66-CD (1999).*

"First analog all-tube orchestral recording in 20 years!" the sticker says on the lid of the CD box. The booklet inside explains further: "This is a pure analogue [sic] recording done exclusively with custom-built triode vacuum-tube electronics. The microphones were arranged in the classic MS configuration." Tim de Paravicini, an audiophile icon since the late '70s, built all the tube electronics, including the A/D. I have a soft spot in my heart for Kavi, as well as for Tim, so it pains me to say that this is not a successful recording. It suffers from too much engineering agenda (all of it retro) and too little pragmatism. A little more trial and error would have helped (maybe there was no time for it). As it is, some of the timbres are quite natural but the tutti are extremely boxy and constricted, with all the orchestra coming out of the middle instead of being spread across the soundstage. The cymbal clashes in certain passages sound especially dreadful. The overall sound has a closed-down, canned quality; you want it to open up and it never really does. The best tube recordings of the '50s and '60s, about which Kavi waxes nostalgic, were considerably better. (Lew Layton, where were you when Water Lily needed you?) Interestingly enough, the last track *(Othello)* sounds better than the rest—did somebody make some adjustments late in the session? I must hand it to Kavi, though—anyone who could persuade the Philadelphia Orchestra and Sawallisch to participate in this audio experiment, especially in the acoustically tricky and microphone-unfriendly Academy of Music, is a diplomat of the first magnitude. I am truly impressed. What's that? You want to know about the performance? The Philadelphians can play this stuff in their sleep.

DVD Reviews

By Glenn O. Strauss, Contributing Editor

The following discs in the DVD format were auditioned using the 96/24 or dts digital-to-analog conversion of the Sony TA-E9000ES and TAG McLaren AV32R processors, as well as the MSB Technology "Link."

Classic Records

Classic is active in many recording formats, including reissues of LPs, original recordings, and DVD-based audio discs using 96/24 technology. Classic refers to the latter as Digital Audio Discs (DADs), which they and Chesky have been releasing as stopgap measures until one of the "super" formats (DVD-A or SACD) becomes a market force.

***Lorna Hunt:** All in One Day. Engineered and mixed by Paul duGré; mastered by Bernie Grundman. DAD 1015 (recorded live without audience in 1998)*

Lorna Hunt is a singer-guitarist who writes songs of reflection and personal relationships. She has a voice with good range, mature expression, and excellent phrasing. This original recording, made in a

1920s-vintage theater and using individual acoustic baffling for the musicians, very nicely balances intimacy and acoustic. For the listener, this means that instruments such as guitar and drums have clean transient attack, while the voice and acoustic bass, which benefit from room reverberation, have body and natural bloom.

While the song material did nor touch me deeply (I prefer more edge to composition), the totality of the sound did. Ms. Hunt's voice is wonderfully rendered, free from the overprocessing of most commercial efforts. To some of the listening panel, it sounded too dry—to those more experienced listeners frequently exposed to natural acoustic sound, all was sweetness and light. As with many female voice recordings, this disc is useful for speaker setup and room treatment.

What about the DAD sound? That's a bit difficult to pin down. Certainly, the delicate acoustic guitar fearherings and percussive attack/decay of the drums on this recording were stunning. There is a wonderful sense of acoustic space, and Ms. Hunt's delicate upper registers are breathy and free from sibilance. I have occasionally heard its equal in the CD format, but seldom. And I think that gets to the point of the matter: with DAD, there is enough headroom and resolving horsepower that superb results are all but guaranteed if the input quality is there. With CD, it takes tremendous skill and care all through the chain to work the magic.

Muddy Waters: *"Folk Singer." Engineered by Ron Malo at Tel Mar Recording Studios, Chicago, IL. DAD 1020 (recorded April, 1964).*

This reissue recording of blues icon Muddy Waters has achieved the status of legend in the audio community, and for good reason. As a towering influence in the evolution of the blues from the Delta style to the urbane Chicago style, no one has had more impact than Muddy. By 1963, he was a major blues stylist and had worked with and discovered some of the best supporting talent then, and now: Buddy Guy, Steve Cropper, Donald "Duck" Lay, Otis Spann, Willie Dixon, Clifton James, and so many others.

"Folk Singer" saw Muddy return to the simplistic style of the country blues: minimalist instrumentation, straight-ahead delivery, and a focus on emotive expression. Here, Waters is joined by Guy, Dixon, and James.

Having owned the LP, the reissue CD, and now the DAD, I was able to do direct comparisons of the CD and DAD (I passed on the LP, since it is mixed so differenrly). Levels were carefully matched. And the result? I preferred the DAD to the CD. It simply had a more you-are-there quality to it. Subtle vocal inflections in Waters's style were easy to identify, such as the way he trails off his voice toward the end of a phrase, only to bring it up a bit at the end. Of course, this was on the CD too, the difference being a subtle but noticeable improvement in resolution in the trailing vocal cues, and a sense of effortlessness. Buddy Guy's guitar had more bite and harmonic richness. Some have reported huge differences in the bass quality, but it was a tossup in my book.

Classic describes its DAD recordings as "Mastertape Sound in Your Listening Room." For once, high-end audio delivers on a promise. This is a classic—no pun intended.

Chesky Records

Chesky has been recording in the 96/24 format for several years. They call their process Super Audio Disc (not to be confused with Sony's proprietary Super Audio Compact Disc). So I guess Classic has DAD, and Chesky is SAD?

Dave's True Story: *Sex without Bodies. "Engineered by Barry Wolifson; mastered by Nick Prout. CHDVD17 (recorded live without audience in New York City, 1997).*

This one too has been around the audio firmament for a while. But it is worth mentioning because it delivers the goods—truly outstanding sonics and much better than audiophile-average musical content.

Singer Kelly Flint and musical driver David Cantor have crafted a wonderful collection of mostly original tunes, and realized them in a hip, NYC-sawy package that has echoes of 1950s coffeehouses but with the wit and perspective of late-1990s urban America. This undertaking could easily have resulted in a pretentious mess; that it didn't speaks to the talent of the participants. I mean it is no mean task to reference Anthony Trollope in a sophisticated, sprightly love song without risking pedantry.

Recorded in a Lower Manhattan church, "Sex without Bodies" evokes the coffeehouse venue I mentioned earlier, albeit a large one. The miking captures a rich reverberant acoustic, which contributes to the sense of liveness that pervades this recording. While this was also evident on the CD, the SAD takes it to a higher level. I hear more of the subtle details that suggest musical life, especially on instruments that render decaying harmonics, such as vibes.

On one of the tracks, there is a buzzing hum during a quiet section, audible on both the CD and the DVD. As a guitarist (electric and acoustic) well experienced in tube-amplifier quirks, I immediately identified it as a guitar-amp glitch. But on the DVD, I was able to hear the very low-level harmonic content of the buzz and identified its source as a tube rectifier heading for tube heaven. Scary.

All in all, a fine effort, and highly recommended without reservation.

Jon Faddis: *"Remembrances."Engineered by Barry Wolifson; mastered by Nick Prout. CHDVD176 (recorded in St. Peter's Episcopal Church, New York City, 1997).*

This is the one that I bring out when someone asks me about the future of digital recording in general, and 96/24 in particular. Each time I play this disc, the hairs on my arms rise up, and I settle into my electronic cocoon a happy man.

Jon Faddis has crafted a nostalgic collection of jazz classics, from Duke Ellington to John Coltrane to Gordon Jenkins. Lovingly arranged and played by a bevy of jazz talents, "Remembrances" is one of those discs perfect for late-night listening.

Two tracks in particular stand out as audio references: "Laura" (from the movie of the same name) and "Naima," John Coltrane's soulful ballad. In each, Faddis's renowned high trumpet register is reproduced with perfect clarity and bite. Bass is reproduced with a convincing sense of power and dimension. But what really sets this disc apart is the ride-cymbal sound. The brushed cymbal sound alone is worth the price of admission, being effortlessly open, microdynamic, and true in timbre. There is none of the leaden undertone sound of conventional CD or the distortion of LP reproduction in this range. Instead, one hears the quintessence of the audio reproduction goal: live sound in the home.

By all means, obtain this disc if you have the technology to play it. Or drop by some night and I'll spin it for you. Bur don't ask to borrow it—you don't want to see this man get mean.

A musical triumph.

dts 5.1 Surround Discs

More and more inexpensive equipment now incorporates dts decoding, not just the exotic front ends.

Brian Wilson: *"Imagination." Engineered by Frank Pappalardo; mastered by Bob Ludwig.* ***DTS Entertainment*** *71021-51018-2-8 (1998).*

Many consider Brian Wilson one of the great talents of the rock 'n' roll era. Count me in that company. Songwriter, musician, arranger, visionary, burnout, recluse, and survivor, Wilson built on the surfing fad and crafted some of the seminal works of the pop repertory

While some of the songs are reinterpretations of classic Wilson tunes, some are new and represent the first commercial releases from Brian in many years. All of the talents that make a Brian Wilson song easily identifiable are in abundance—the catchy tunes, the imaginative bridges and choruses, the rich arrangements, the key and timing changes, and the always unexpected selection of instruments. All there.

So what went wrong here? Well, the sound. There is a nasty, edgy sound to Brian's voice that cuts through the mix, but also cuts through the tweeters, the listening room, and one's ears. Instruments are similarly processed, overprocessed, and equalized/compressed into a hashy porridge of sound.

There are moments of greatness here, although Wilson's sophomoric and at times whining lyrics are not engaging. But just as some of Mozart's comic operas have silly and dated librettos, Wilson's musical talents rise above the spoken or sung word.

And the Wilson/Mozart comparison is more than skin-deep. Both are acknowledged for their ability to take a simple tune or motif, add seemingly simple orchestral lines, and achieve a result that is somehow bigger than the sum of its parts. Genius? It's as good a word as any.

If you are a Wilson fan, this is worth the price of admission; on strictly sonic terms, it doesn't make it.

Gaudeamus: *"Sacred Feast." Engineered by Tom Jung and Tom Bates; recorded at Trinity College Chapel, Hartford, CT. Music as Software* ***(dmp)*** *MAS CD-805 (1998).*

If the Brian Wilson dts disc was a disappointment, this recording of a men's choir a cappella was an unexpected delight.

Tom Jung used a modified Decca Tree microphone setup for this recording. It paid off. Both the direct sound from the choir and the early- and late-arrival reverberant soundfields are wonderfully captured. But that is also true of the stereo CD version, which Peter has reviewed above. It is in the 5.1 surround mix where dmp has truly excelled.

Most surround software has gone overboard in the mixing to the center channel, and especially the surround channel. Other than a few Telarc dts recordings and John Eargle's work for Delos, engineers of classical discs have tried to draw too much attention to the surround channels at the expense of the L/R channels. The resulting "Hello, I'm here; I'm a surround recording—listen to me!" effect has been far less than natural and has done little to advance the acceptance of the format by serious music lovers.

This one gets it right. The surrounds and center are artfully used to embellish the L/R channels, and add the acoustic ambience and dimensionality that surround sound promises but seldom provides. In a properly balanced surround system, the sense of acoustic space is huge. One can almost get the feel of the wood, plaster, stone, tile, and even the air of the chapel.

Much has been written about dts's compression algorithms, and the general tone of the audiophile community is that it is good for films, but not nearly resolving enough for serious musical enjoyment. And those audiophiles are already blase about 96/24 recording, screaming for 192/24 and beyond! As Peter mentions in his SACD review in this issue, with results like these one can make a strong case to question the need for super audio formats.

Still, I am more friendly to the notion of extended-resolution digital formats, so I will wait until I have a DVD-Audio 6-channel data stream feeding a processor the same music as recorded on this disc, to make a sonic comparison.

Until then, this mighty disc pleases mightily.