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CORRECTION

Cynthia Wenslow was the photographer of Photo 1 in our Burning Amp show report (Jan. '09, p. 39). We regret the omission and apologize for any inconvenience.—Eds.

COMPUTERS AND ANALOG AUDIO

Since digital audio—and converting analog audio to digital—is such a hot topic lately, I just wanted to share a few observations and suggestions on the subject.

1. If you are planning to digitize your record collection, give serious thought to what you use for the conversion. The sound cards built into many computers (especially the on-the-board ones) are sometimes *really* bad—worse than you can imagine—hissy, noisy, high distortion. If your recordings all seem noisy, or have little twittery noises, don't waste time trying to fix the unfixable—you can get a pretty nice sound card for a desktop machine for about $30. Try the Sound Blaster Audigy SE, which has an odd quirk of using the same plug for line and mike in and switching them via software, but is relatively quiet and clean, sounds quite nice, and costs less than $30 at Wal-Mart or via mail-order.

2. Desktop recording software, especially the commercial applications such as Adobe Audition (a *very* nice program), has many features and requires a lot of computing power. If you're using an older machine that isn't all that fast, you may not be able to record audio with some programs (editing, because it is not done "real time," will just take longer on a slow machine—recording, however, will skip and hiccup and do all sorts of evil things). The answer is to use *less* powerful software when recording. While I was trying to do some recordings for a friend on an old 866MHz Dell machine—with the Audigy SE card, Adobe Audition couldn't record at all without continual glitches and hiccups. The solution was to use the simple little recording utility that came with the sound card to record—which worked perfectly every time—then load the files into the good editing program to edit them afterwards.

3. Generally, real-time (as in audio recording) is easier on a fast machine, and, with slower machines, you must be careful to not "distract the machine" while actually recording.

You can use a separate drive partition for recording because the C drive *always* becomes fragmented. An external USB drive may be a good answer, *but* make sure that your machine has the newer USB 2.0 high-speed ports. Older (pre-2001) machines often have the slower USB 1.0 ports, which are supposedly fast enough for audio, but are really borderline. Adding a second internal drive would then be a better bet, or you can update the USB ports by adding a USB 2.0 card for about $15.

Of course, try and ensure that other programs aren't running, *especially* antivirus and security programs. If they are, be sure to switch off the "live scan" function while recording, which slows the machine down a *lot* even when it's not scanning the drive. (You can disconnect the machine from the Internet while recording if this makes you uncomfortable.)

If you have an old machine that's not being used, it might make sense to dedicate it to audio recording. Partition the hard drive (or add a second one, which doesn't need to be big by today's standards), leave off all extra software (such as antivirus programs), and install a decent sound card. I have an old Dell 866MHz box set up that way with the Audigy sound card, a 20GB hard drive divided in half—so half is a clean 10GB "music" partition—the Audigy (free) recorder to record, Adobe Audition to edit, and nothing else running on it. It works *very* well.

4. Sometimes it's difficult to figure out what you're looking for, and what it's called. You may find that a computer store, an audio store, and an audio "studio equipment" store each sells something that *does* the same thing, but they each call it something different, and the prices are wildy different as well. This is especially true for computer/audio gear.

For example, you may want to purchase a good-quality analog output from your laptop—the one with the not-so-good audio card. You can buy a "USB Headphone Amplifier" on eBay for $85; a line-out equivalent for a similar price; a "USB Sound Card" from a computer outlet for anywhere from $6.95 up (many of which also take analog IN and convert it to digital); or a "USB 2-input/2-output Digital Audio Interface" by Behringer from a studio-audio outlet for $29 that does both input and output *and* provides a digital output for connecting your own outboard DAC (a feature that generally costs about $100 from a "home audio" device).

Behringer also makes a "headphone amplifier" (AMP800-MINIAMP), which has two inputs and four headphone outputs—each with a separate level control. Real handy if you connect it to your computer and connect your powered speakers to one "headphone out" and your headphones to the other. It even includes another input (with level control) for your iPod, which you can send to any of the four outputs individually. (Yes, it might be more fulfilling to build a better one, but, for $39, you can't even buy the case and connectors, and, if you need it today, you're sunk.)

For that matter, people spend serious money for "iPod speakers" when iPod speakers, "self-powered studio monitors," and "computer speakers" are all the same thing—speakers that take line-level inputs. Cheap "computer speakers" make great iPod speakers, and cheap "powered monitors" make even better computer speakers.

5. Grounding. Computers still need to be grounded. Even a laptop, running on batteries, may be susceptible to noise pickup. If so, finding a piece of actual metal on it and connecting it to an earth ground—like the ground pin on the outlet strip—will help (sometimes the little screw fittings that you use to secure the VGA connector are actually connected

to the metal chassis when nothing else seems to be.

6. Digital recorders. The latest of a new generation of portable recording devices, the Zoom H2, can record stereo or surround sound digitally to a standard camera-type SD memory card (which is now incredibly cheap). It has four *built-in* microphones. Because the recordings are already digital, you don't need to convert them, and the quality is very good. It also has microphone and *line* inputs, so you can use it to digitize your line level inputs—such as from your turntable.

It can record low-quality MP3s to four-channel 48kHz/24 bit files, has several limiters and filters which can (optionally) be invoked, and runs for about four hours on a pair of AA alkaline batteries. You can also connect it to your computer and use it as a digital sound card and/or digital microphone, and it has threads to fit a standard camera tripod mount. You can also take its microphone output from the line out and use it as a microphone with built-in preamp. And it's about the size of an electric razor and costs about $200. There are several competing products which claim to be even better (for a bit more).

7. Last, digital sound in general clips *hard* when it clips. When you record digitally, you can usually afford to record at lower levels than you might be used to, because most converters are very quiet. This is good, because, unlike analog systems, digital systems clip *absolutely* at 0dB (or whatever their "0dB" level is). There is no graceful rounding of waves, or gradually increasing distortion.

Most digital devices are perfectly clean at -0.1dB and cut everything off flat at zero. So, when using a sound card or digital recorder, you want the levels to be good, but you *never* want the peak indicators to flash—not even occasionally. Reduce the level until they don't if you want to stay clean. Luckily, most digital recorders have peak indicators that are *really* instantaneous, responding to a single sample that clips.

Also, if you're new to digital recording and editing, be *sure* to read the book for your editor—there are all sorts of cool new features that simply didn't exist with analog (check out "normalize" and "noise removal" and, in Adobe Audition, "edit in spectral view"). Before you buy a cheap editor, or settle on a free one, download a demo copy of Adobe Audition and decide whether you can really live without the extras. Most programs even include the ability to make perfect test tones, filtered noise, and sweep tones—just by punching in what you want (such as "431Hz sine at -3.26dB for 13.4 seconds").

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SIMPLE PROTECTION FOR SPEAKERS

When I read about David Mayfield's speaker mishap ("Speaker Setback," Xpress Mail, Jan. '09, p. 42), I thought I'd pass along the following simple suggestion. By connecting a 0.5A fast-acting fuse in series with each of your speakers, you can protect your speakers from power company mishaps, like in David's case, and from solid-state amplifier failures which can result in destructive DC voltage at the speaker terminals.

Using the Radio Shack 270-1281 in-line leaded fuse holder makes this quick and easy. Normal-to-loud listening levels won't blow fuses, whereas destructive AC or DC voltage blows fuses instantly. It's a great alternative for speaker owners who don't wish to bother with complicated speaker protection circuits.

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HORN OPERATION

In response to Mr. Kolbrek's response to my letter ("How Horns Work Revisited," Dec. '08, p. 18):

I would tend to agree that my dismissal of the horn equation may be a bit too absolute for every case, but for what I do, it has not shown to be useful. In some circumstances I can see where its simplicity might be attractive.

Quite recently I had a thought: if we have an exact formulation of a "horn," namely the OS waveguide, for which solutions are available, then by simply plugging the OS contour into the horn equation a reasonable comparison of the accuracy of the two approaches can be made. I tried this and the results were quite interesting. The mathematical details of solving these equations will need to be forgone for brevity, but the conclusions are straightforward and simple.

The horn equation appears to break down right at cutoff, and it is not hard to see why. It is right at cutoff that the wavelengths begin to exceed the size of the device itself. This means that in Morse's original assumption limitations

$$\left|\frac{d}{dx}\sqrt{S_0 \cdot e^{m \cdot x}}\right| << 1.0$$

it appears that we should actually compare the rate of change to the wavelength. This makes perfect sense because it is actually the rate of change of the boundary with respect to the wavelength that determines how "fast" the boundary changes.

In the horn solution of the OS waveguide the answers are completely wrong right at cutoff, but become progressively better above cutoff. And yes, a clear cutoff is predicted in the horn solution that does not exist in the exact solution. The "loading" of the OS goes to infinity at cutoff and becomes imaginary below this point, almost exactly like some other horn contours do.

Above cutoff it is found that the relationship θ/***a***, where θ is the wall asymptotic wall angle and ***a*** is the throat radius, plays an identical role as the flare rate ***m*** does in an exponential horn. In essence, as far as horn theory is concerned, the OS waveguide and the exponential are almost completely equivalent contours.

In this light it appears that the horn equation is not very good at predicting the loading of a device near its cutoff, but, of course, because all devices become the same at higher frequencies, it works fine in that region (as far as the impedance goes).

In regard to my claim that virtually all contours for practical devices will result in equivalent performance, I submit the following simulation (**Fig. 1**) of the same driver on three devices, conical, exponential and oblate spheroidal, with identical input and output areas and length.

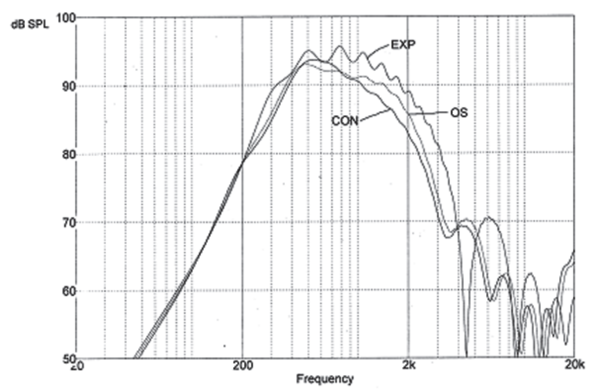

FIGURE 1: Driver simulations for conical, exponential, and oblate spheroidal devices.

This simulation uses a finite (conical) element simulation for the contours, *not* the horn equation or the wave equation. It is thus more accurate than the horn equation but less accurate than the exact wave equation (which only exists for two of these devices). It is the most accurate at the lower frequencies where, as I have discussed, the horn equation is losing accuracy.

You can see that there are differences, but these differences are not such that one device is clearly superior to another at the lower frequencies where the model is the most accurate. The exponential horn appears to have a slight gain advantage, but it also appears to have a highly resonant loading. The flare rate for the exponential in this case is not very great, which is why its efficiency appears higher, but it also beams more with substantial lobing, although at these higher frequencies the directivity model is becoming suspect. At the lowest frequencies and near cutoff is where they all tend to look about the same, and this is precisely where the horn equation says that they should be the most different. The horn equation just does not seem to give useful results at low frequencies because, as I have said, all contours appear to be converging on the same answer, and it fails at higher frequencies because its plane wave assumption is invalid.

In conclusion, let me just say that throat impedance and efficiency are not irrelevant in all situations, but they are not nearly as important or as different as the horn approach would lead us to believe. (Nonlinear distortion performance of horns and waveguides is not at issue here because neither approach offers anything in this regard, so I am not sure what Mr. Kolbrek is referring to, but I would suggest a review of the several papers that my colleague and I have done on distortion perception which shows how the horn is very unlikely to be a significant factor in the "nonlinear" sense, although its HOM (high-order modes) performance appears to be critical.) The differences in various contours are predominately at the higher frequencies where the horn equation offers no insight into performance.

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CLASS D AMPS

I just wanted to add a few more comments to Bruce W. Brown's excellent article about Class D Amplifiers in the November 2008 issue (p. 22).

First, I do want to dispute the oft-repeated claim that "the design of class A amplifiers produces the least distortion." From an engineering point of view this is a bit of a simplification, and really only applies to relatively simple transistor and tube circuits. The premise carries over when feedback is added into the picture because "regular" negative feedback serves to reduce distortion—so, the less distortion you start with, the better off you are. Class B and higher types of circuits are prone to distortion because they "hand off" the signal between multiple devices, and because devices are rarely—if ever—identical, or identically opposite enough for distortion to cancel exactly; this handoff process tends to cause distortion.

If you look at the individual parts in a class D amplifier, they each produce nearly 100% distortion, because, by design, they are not operated in a linear manner at all. If, however, you look at the overall circuit as a single "device," then a good PWM amplifier could easily produce less distortion than a good class A amplifier. In theory, a good PWM amplifier could also be built without using feedback, although the sort of folks who design class D amplifiers generally aren't superstitious about avoiding feedback. The main difference is that the transfer characteristics of tubes, transistors, and FETs are a property of the device itself, while those of a class D design are properties of the circuit and are quite divorced from the properties of the devices used in the circuit.

For those not interested in engineering technicalities, you should simply accept that class D amplifiers cannot be directly compared to linear amplifiers (the analog kind) at the circuit level—and trying to do so is pointless. For example, some class D designs are sensitive to power supply noise and fluctuation, while others are entirely immune to it.

One fair generalization is that class D amplifiers must be more complex in order to produce satisfactory audio performance—nobody has yet demonstrated one using four or five parts that sounds anything other than awful. But, if you count an integrated circuit with a bunch of parts inside it as "one part," then there are many single chip class D amps that far outclass most class A SET triode amps.

There are many interesting class D amplifiers available—and some of them are amazing bargains. The Bel Canto model that Bruce is hoping to audition uses a chipset made by someone called TriPath (now maybe out of business, as it turns out). There are several models using the same or similar chip sets which are available for much less money.

41Hz (www.41hz.com) offers several nice-looking class D amplifier kits (using TriPath chips) of various power levels and configurations—in both surface mount and throughhole versions (you can do digital with non-SMD components, it just takes more design effort). I have ordered two and they appear to be well designed—but I haven't gotten around to building them yet.

Trends Audio (www.trendsaudio.com) sells a very small amplifier called a model 10.1, which puts out about 10W/channel (WPC) using a TriPath chip—for about $150. It runs on a 12V 4A power supply (included), or can be run on a big 12V battery or car voltage if you like. It sounds remarkably good, absolutely quiet, and very clean. The build quality is also excellent.

I haven't done any serious comparisons, but, when connected to a pair of vintage Advent Smallers in place of a 190 WPC Hafler power amp, the little Trends amp had cleaner midrange and the bass sounded clearer and obviously less muddy. It was also able to reach reasonable listening levels (although, obviously, the Hafler would play louder).

This is very impressive considering that the Advents have heavy woofers known to require solid damping and serious power to work well, and less-than-impressive midrange—and considering that the Trends amp is less than 1/10 the power of the Hafler and much less than 1/10 the weight or size. The improvement in sound was not at all subtle.

For the budget-minded, there was an amplifier based on the same or a similar chipset available until recently for $29 (the Sonic Impact T-Amp). It is small, very plastic, has really cheap thin binding posts, and will produce about 2 WPC using eight AA batteries or about 10 WPC using an external—and not supplied—12V power supply (plug-in, battery, or car). The entire circuit inside is about the size of a fat thumb—including the sizable heatsinks that it doesn't have and doesn't need—it doesn't even become warm under normal use. It sounds remarkably good when running on external power (in my opinion, and those of many reviewers, better than the few single-ended-whatevers I've heard). It even sounds quite good at low listening levels running on AA batteries—a set of which will last eight or ten hours—which makes it a great laptop or iPod amp. The company has discontinued this model, but they now sell a more expensive model of basically the same amp, and the original occasionally turns up on eBay (now not cheap) or in clearance bins.

As with many modern products, the sound quality of class D amplifiers is market driven. There are some out there that sound very good, and others that do not—and, obviously, in many cases the size and weight are the major considerations in the design. The point is that they are quite capable of sounding very good if well designed and implemented. It would be a serious mistake to assume that a good class D amp may not equal or top a good amp of any other type.

Unfortunately for DIYers, class D designs tend to be quite complex, both because of the circuit complexity and because of the design and fabrication requirements due to the high frequencies involved. Even the chipsets tend to be quite fussy about circuit layout and component specifics. Unless your high-frequency design skills are up to date, it's probably safer to stick with prefabricated PC boards if you set out to build one. (Check out the user forums on www.41hz.com for a good idea of the design considerations and pitfalls common to class D amps.) In general, many parts in certain circuit locations must be of specific types because of the high-frequency requirements, and so you should not subsitute parts unless you are very familiar with the design requirements.

On the upside, most of the chip-based designs are quite insensitive to power supply variations and noise beyond having sufficient power of reasonable quality available, so power supply design is usually simplified. (Many of the 41Hz models include a simple but sufficient power supply on the amplifier board—of the "four diode bridge and a capacitor" variety.)

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CD PHENOMENA

In reference to Keith Levkoff's excellent, somewhat neophyte-oriented, treatise on CDs ("How CDs Work," Xpress Mail, Dec. '08, p. 42) and proper lambasting of "magic" potions to improve audio playback quality of the same, I would like to add a few observations and experiences gathered over the last 35 years of audio engineering.

In the mid-70s, when CDs were relatively new, and at my first retail hi-end audio salon work experience, a sales rep paid us a visit, flogging a nifty little green magic marker ($79 retail). One was to apply around the outer circumference of the disc, thereby removing reflections from the polished edge of the new CD and magically improving the sound. . . somehow. When he was unable to explain scientifically how this might be, he was sent on his way with much derisive laughter from me and my skeptical colleagues. We peeled the label off the sample and discovered a Schaeffer No. 8 green marker. We actually tried it, just to be fair, in simple blind tests and had trouble breathing from laughter, and washing the green ink from our hands.

On a more serious bent, many years later, as the head service engineer for the Canadian Linn distributor, I experienced very strange and repeatable CD hardware phenomena, about which I am still uncertain. As readers may be aware, Linn makes some very respectable and spendy CD machines. While servicing a big-dollar CD12—arguably a very fine Aston Martin-like player—I came across an odd phenomenon during repair. The laser "sled" was obviously faulty, with no laser focus activity on power-up. I replaced it, having a dozen brand new ones in stock, and cleaned and lubed the mechanism. Note that Linn units have very few, if any, adjustments required or available, as, say, in adjustable trimmers, for focus offset, focus gain, and so on, unlike some other machines.

During my final listening test, Diana Krall's drummer's snare brush *only* sounded nasty. Like crap, actually—hard and clearly distorted in some way. *Every other instrument* on this track, including her voice and cymbals, sounded perfectly fine or as good as my musical memory said. This was a very clean shop test disc, used and heard daily. The shop listening equipment was very modest, but the bad sound was plainly obvious in spite of this, and also later, with a room full of experienced people. They were sent on their way.

As an experiment, I replaced the laser sled with a new one. Voilà! Glorious! No bad snare brush sound. I repeated this A-B-C several times with the same results and the same listeners. The "eye pattern" (the high-frequency RF signal from the working laser outputs) on the scope was perfect-looking *in every case and aspect*. This confused me, to no end.

Caveat: Since the 70s, I had probably serviced or replaced literally thousands of CD laser mechs of every brand on the planet, and as such, have a pragmatic disdain for audio "voodoo," or attempts to legitimize marketing audio BS, in any way. That said, I encounter a head-scratcher once in while. If it's played back with all 16 or 24 bits intact and synchronized, it is what it is, or is it? Laser jitter? Any similar technical experiences out there?

The "eye-pattern" previously mentioned can be compared to the signal coming directly off a tape recorder head, if you like. It's measured at the source, or as close as is practically measureable. In case you weren't aware: When you burn one on a typical PC or Mac, the level *coming directly off the burned disc* is approximately 60%

that of a commercial stamped audio CD. If your machine's servos and so on have no problems tracking at this actual lower recorded level, it will sound normal.

Where this issue typically shows up is in your crappy car CD player where the focus error/tracking/offset correction servos are already working overtime anyway, from bumps and vibration, due to driving on the road. Don't necessarily accuse your wives or kids of throwing your jazz CDs on the car mats when your car copies start skipping. On the other hand, it's a great excuse to not be willingly subjected to that vile and pernicious "Mamma Mia" soundtrack.

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WT3 PRINT OUTPUT

In the "Dayton Audio WT3 Woofer Tester (Nov. '08, p. 32)," G.R. Koonce notes, "The once missing file option seems to be the ability to output the graphs to files as pictures. This would be useful when using the WT3 to prepare material for publication. . ." I would suggest trying the "PrintScreen" key as follows: hold down "Shift" + "PrintScreen"; or "Ctrl" + "PrintScreen," or "Alt" + "PrintScreen." Any of these actions will allow the screen picture to be "pasted" into a document, by holding down "Ctrl" + "V" or right-clicking the mouse and selecting "Paste." You can thus copy the screen picture onto a WORD document.

Or, better yet, copy it into a PAINT document. Then place the mouse-cursor over the "Select" icon (which is directly below the "Edit" choice on the menu line). Position your mouse "Cross-Hairs" at the desired point on the picture to begin cropping the desired section of the picture. Just hold down the left-click button on the mouse and drag to the opposite end of the picture in order to crop the desired section. Let go of the left-click mouse button. Next, hold down the "Ctrl" key + "X" in order to "cut" the cropped section. Then switch over to a WORD document and hold down the "Ctrl" key + "V" in order to "copy" the cropped section into the WORD document. Presto! Instant picture!

The original "screen dump" (**Fig. 1**) includes all the top and bottom menu items, which are extraneous and unwanted. Cropping could result in the text being cut off or stretched. Hope this is helpful.

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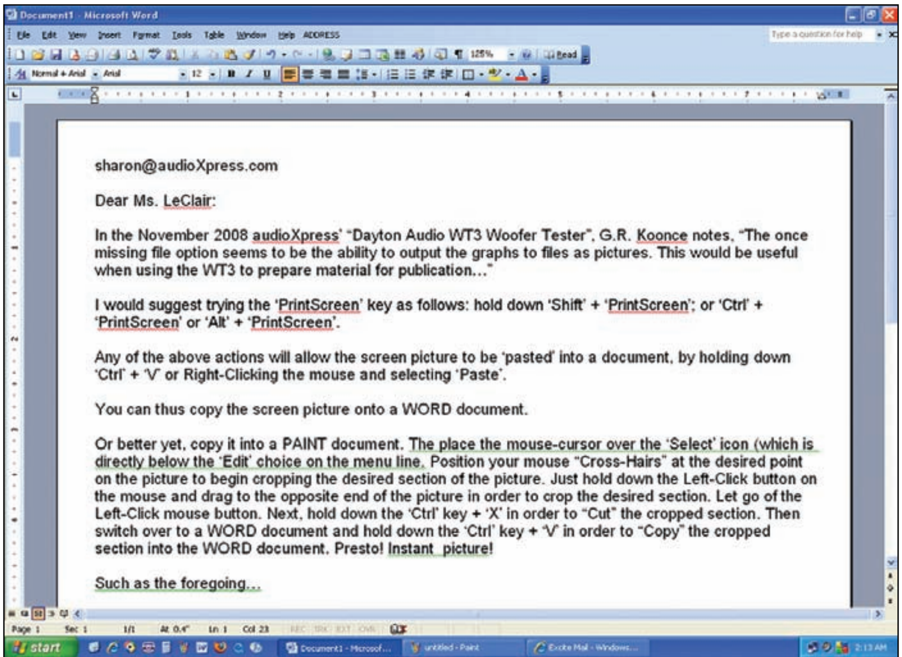
aX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Dear Ms. LeClair:

In the November 2008 audioXpress' "Dayton Audio WT3 Woofer Tester", G.R. Koonce notes, "The once missing file option seems to be the ability to output the graphs to files as pictures. This would be useful when using the WT3 to prepare material for publication..."

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Such as the foregoing...

FIGURE 1: WT3 screen dump.