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By Bill Fitzmaurice

The XF210 Electric Guitar Cab

For best onstage performance, take note of this designer's latest offering.

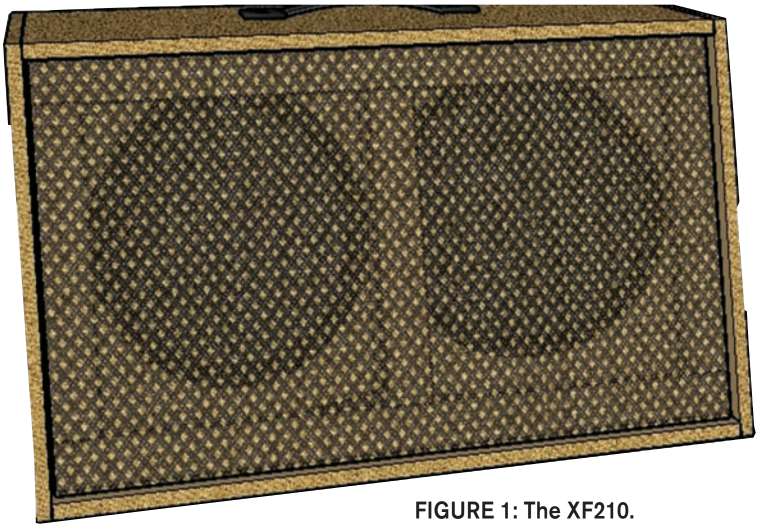

FIGURE 1: The XF210.

Electric guitar speakers are intended to be loud, yet most of them share an incongruous common trait: you can't hear them. To be more exact, you can't hear their high-frequency output unless you're directly on-axis. There are a number of reasons why, foremost being that they use large wide bandwidth (to 5kHz) drivers—10 and 12″ being the most popular—that have very narrow dispersion at high frequencies. To compound matters they're usually placed on the floor just a few feet in back of the player, so high frequencies pass by the player generally around knee level.

CABINET ALIGNMENT

Even worse, to cosmetically "look right" beneath an amplifier that's 20 to 24″ wide, dual driver versions have drivers placed horizontally. This narrows the horizontal dispersion angle in those frequencies where the drivers are mutually coupled, and causes comb filtering in the frequencies too high for coupling to occur. In an audio engineering course on how not to build a speaker, the typical electric guitar cabinet would be around the top of the list.

The simple solution is to vertically array the drivers, which would widen horizontal dispersion, eliminate combing, and put the source closer to the player's ear level. But the resulting tall, thin cabinet would "look funny" with the amplifier overhanging both sides. Guitar players themselves have never had a problem with "looking funny," but they do draw the line when it comes to how their gear looks.

Enter Plan B: the cross-firing 2 × 10. I call it XF210 (**Fig. 1**) for short. While many guitar cabs are angled back a few degrees, the XF210 has a rear tilt of 15°, so the player can actually hear what he's playing. The two drivers also cross-fire inward at a 20° angle. This minimizes the center-to-center spacing of the drivers for reduced combing, and widens high-frequency horizontal dispersion by up to 40° compared to a flat baffle.

Guitar cab alignments tend toward two varieties—sealed and open back—with most players preferring one or the other. To appeal to both camps, you can build the XF210 with a removable rear panel that allows it to be either sealed or open.

Drivers come in many varieties, but by far the overwhelming choice of most manufacturers, and thus most players, is Eminence. To choose one that suits your needs, go to www.eminence.com/guitar.asp?speaker_size=10.

CONSTRUCTION

The cab is made entirely from ½″ plywood. While you may opt for Baltic birch, it's not always easy to find and seldom inexpensive. A void-free grade of softwood plywood of at least five plies is easier to find, and quite adequate for the job.

All joints are glued and fastened. My preference for glue is polyurethane base construction adhesive, caulking-gun applied, because it's inexpensive and expands to fill voids as it cures. Polyurethane woodworking glues are also very good, but tend to be very expensive. Traditional woodworking glues will work, but require care to be sure of airtight joints. Fasteners may be drywall or furniture screws, ribbed paneling nails, or 16/18 ga brads installed via pneumatic nail gun.

Start by cutting out the bottom. **Figure 2** shows the underneath/outside view; the front edge is cut at a 15° angle. **Figure 3** shows the inside view, with reference lines drawn at the joints of the baffle half leading edges. The upper and lower back pieces are cut per **Fig. 4,** the sides per **Fig. 5**.

Attach one side to the bottom. While fastening, hold the two pieces in alignment with a 2 × 2 caul and clamps (**Fig. 6**). Attach the lower back to the assembly in the same fashion (**Fig. 7**).

The mirror-imaged baffle halves are cut in two steps: first, as shown in **Fig. 8**, cut the driver hole diameter per the specification in the driver datasheet, and then remove wedge-shaped portions per

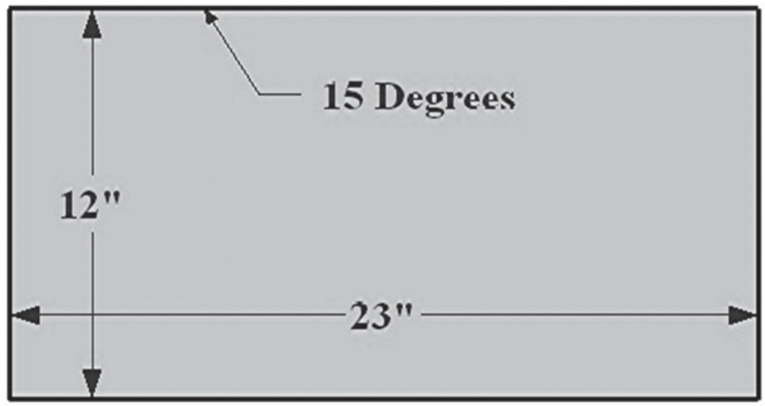

FIGURE 2: Bottom, outside view.

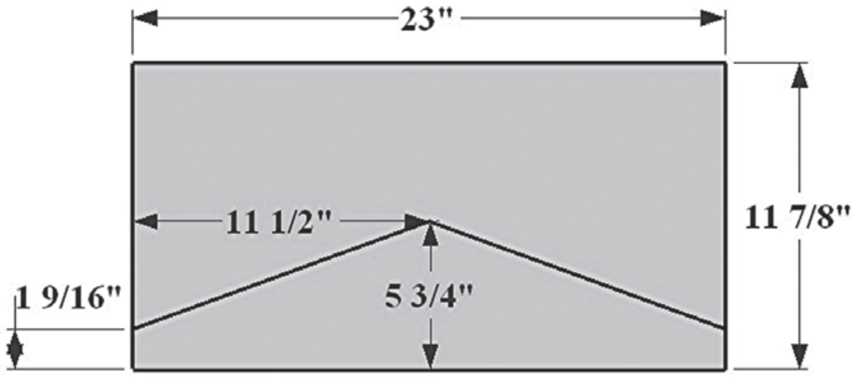

FIGURE 3: Bottom, inside view.

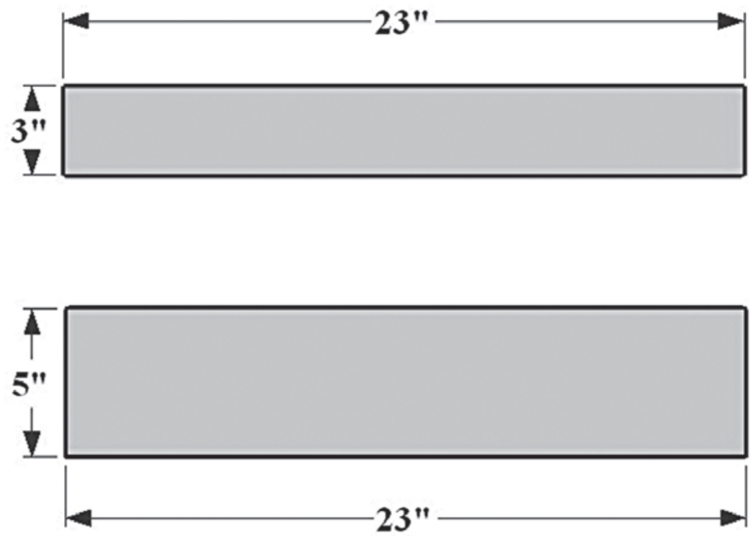

FIGURE 4: Back, upper and lower.

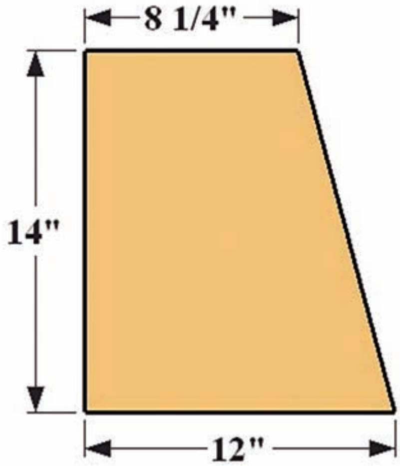

FIGURE 5: Sides.

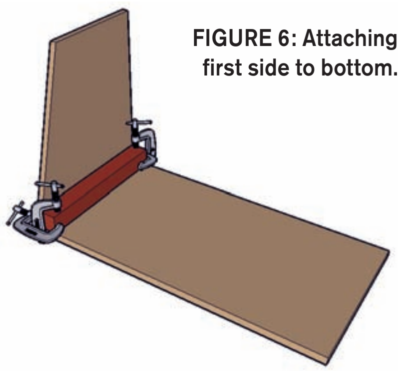

FIGURE 6: Attaching first side to bottom.

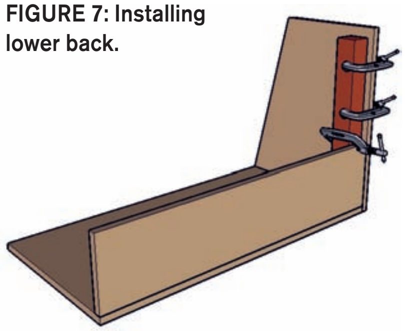

FIGURE 7: Installing lower back.

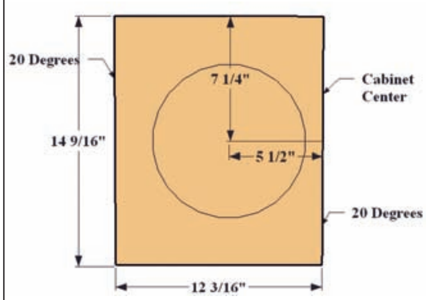

FIGURE 8: Baffle first cut.

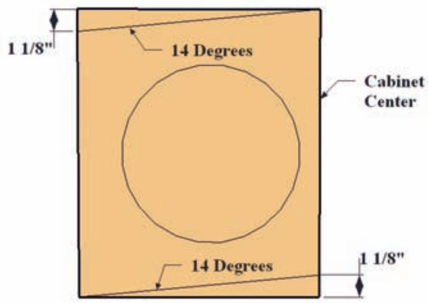

FIGURE 9: Baffle second cut.

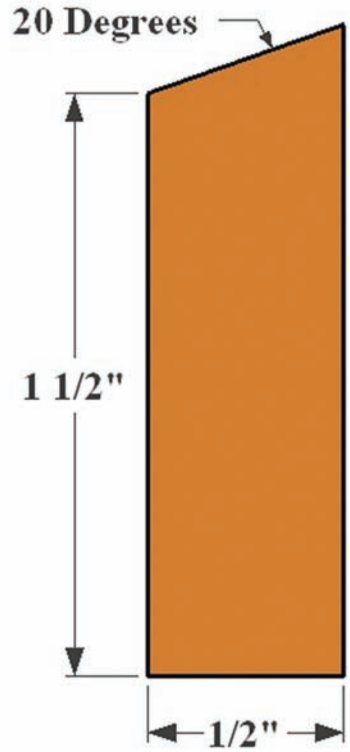

FIGURE 10: Baffle mounting jig.

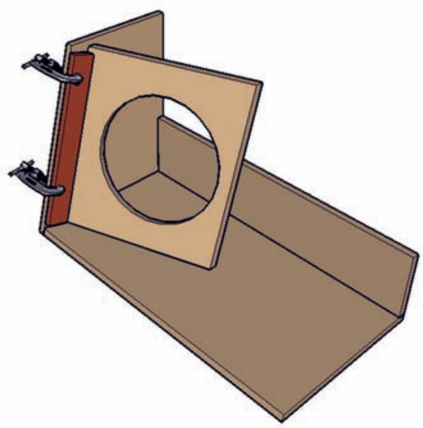

FIGURE 11: Installing first baffle half.

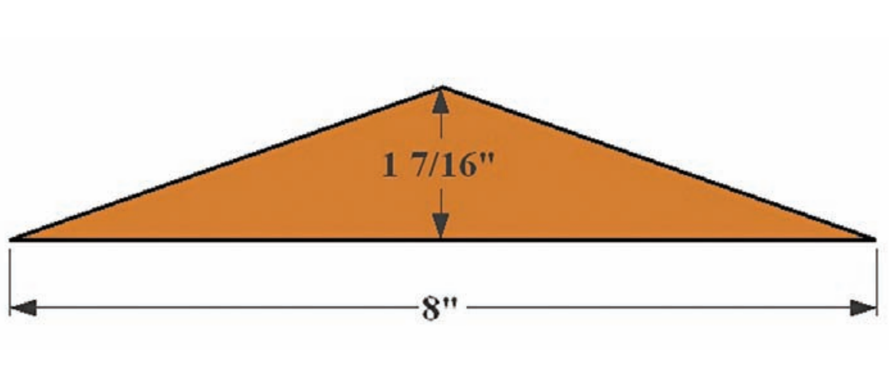

FIGURE 12: Baffle installation brace.

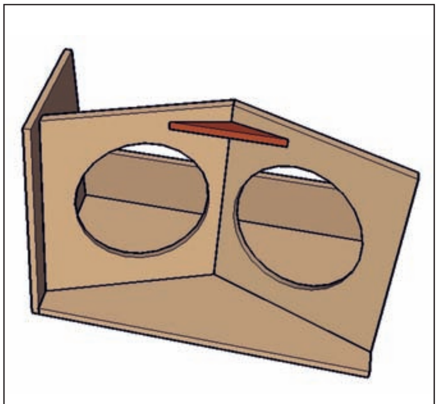

FIGURE 13: Installing second baffle half.

Fig. 9. You may attach the drivers with either screws or bolts and T-nuts. Insert a driver through the mounting hole, marking through the frame the locations for pilots.

Remove the driver and drill the pilots. Repeat with the other baffle half. Install T-nuts from the rear of the baffle, pulling them through with a bolt driven by hand.

Cut a 12″ long baffle mounting jig per **Fig. 10**. Clamp the jig to the leading edge of the cabinet side, holding the baffle half-tight to it while driving fasteners through the side (**Fig. 11**). Align the baffle half to the reference line on the bottom while driving fasteners through the bottom.

Add the second baffle half. A baffle installation brace (**Fig. 12**) screwed to the junction of the two baffle halves (**Fig. 13**) holds them in alignment. Use waxed paper between the brace and the baffle halves to prevent the brace being glued to them. If you wish, use two installation braces, at the top and bottom of the baffle assembly.

Install the upper back, with caul and clamps holding the parts secure (**Fig. 14**). Add the top (**Fig. 15**) to the assembly (**Fig. 16**), then remove the caul and clamps and the baffle installation brace. Add the second side (**Fig. 17**).

Cut grille backer sides and top/bottom per **Figs. 18** and **19**. The backer sides install tight to the baffles; the backer top and bottom connect them across the top and bottom per **Fig. 20**. Cut three grille verticals and the grille top and bottom per **Figs. 21** and **22**. Assemble them (**Fig. 23**), being sure the top and bottom angles face the right way to fit onto the cabinet. You

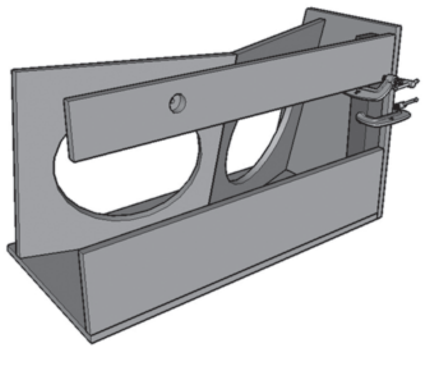

FIGURE 14: Installing upper back.

can assemble the grille with glue only, clamped until it sets, but if you have a biscuit jointer it's the ideal tool for this job.

For a sealed or convertible cab, install inch-wide flanges around the back opening (**Fig. 24**) and cut a back door (**Fig. 25**). Omit these for an open back. Cut a hole in the upper back for a jack or jack mounting plate.

DRESSING IT UP

Apply a finish. While carpet is a durable option, most players will prefer the vin-

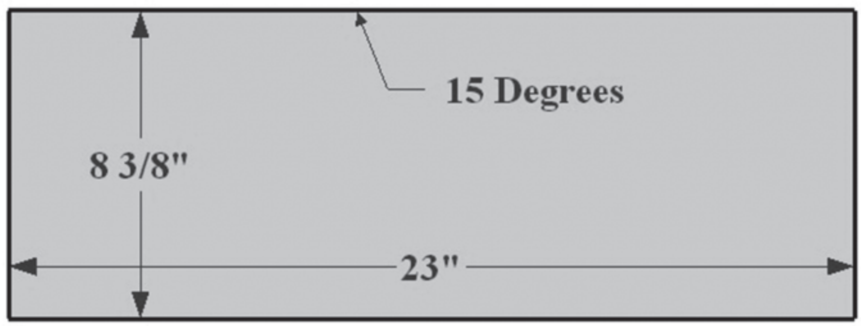

FIGURE 15: Top, inside view.

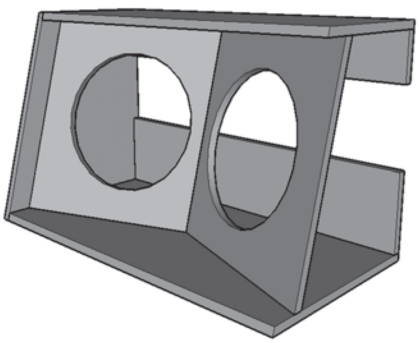

FIGURE 16: Top installed.

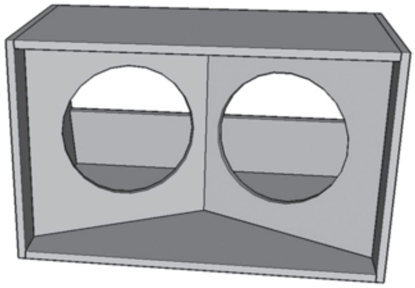

FIGURE 17: Second side installed.

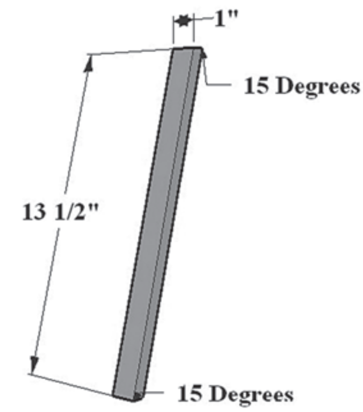

FIGURE 18: Grille backer sides.

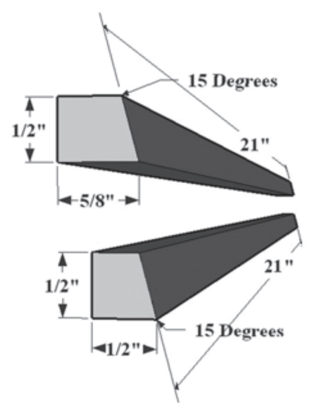

FIGURE 19: Grille backer top/bottom.

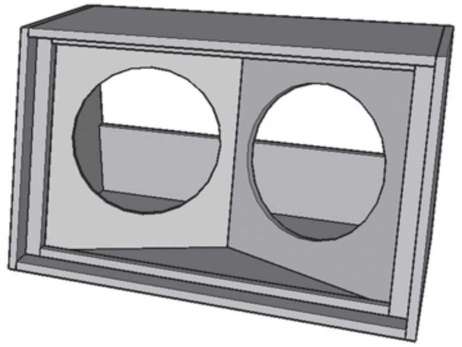

FIGURE 20: Grille backers installed.

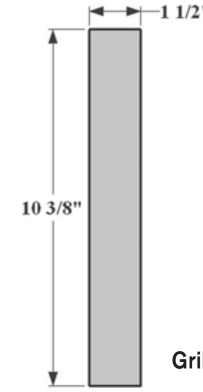

FIGURE 21: Grille uprights.

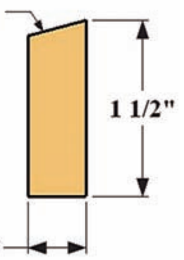

FIGURE 22: Grille top/bottom.

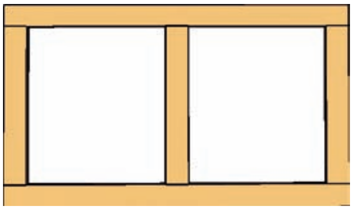

FIGURE 23: Assembled grille.

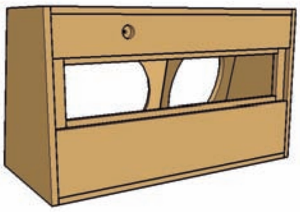

FIGURE 24: Flanges on the back door opening.

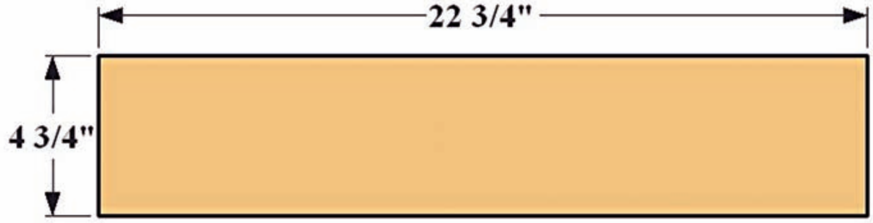

FIGURE 25: The back door.

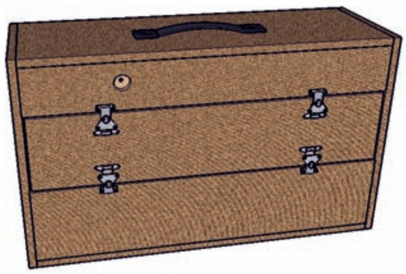

FIGURE 26: Back door secured with butterfly latches.

Drivers, grille cloth, Tolex, and all other hardware are available from Parts Express (www.partsexpress.com).

tage look of Tolex or tweed. Attach grille cloth to the grille. Line the inside of the cabinet with 1 to 1½″ thick acoustic foam or open-celled polyurethane mattress-topper foam, secured with staples or adhesive. Install handles and protective corners as desired.

Install grille-securing hardware to the grille and backers, or use screws with finish washers driven through the grille face. Install the drivers, the grille, and the jack, wiring the drivers, in parallel, to the jack. For a sealed version screw the door in place. For a convertible version secure it to the cab with butterfly clamps (**Fig. 26**), with foam weather stripping on the flanges to prevent vibration. *aX*

A Really Simple **PAS Modification**

With this mod, you'll discover why this classic unit is a favorite of audiophiles.

The ubiquitous Dynaco/Dynakit PAS must be one of the most popular tube preamps in history. It also must have drawn the most modification "ink," with at least two articles in Audio Amateur Publications: *Audio Amateur*, "PAS-MOD Mk.III," 4/82, and *Glass Audio*, "Spice and the Art of Preamp Design," 3 and 4/97. It draws good press and popularity for many reasons, including the fact that it sounds pretty darn good in "stock" form. The main problem I always had with it was that it was prone to noise and hum; didn't sound flat with the tone controls centered; and the tone control range was *waayyy* too much—could induce turntable rumble with the bass moderately up. Please note that this modification does *not* apply to the last version of PAS preamps manufactured, which I believe were called the -3X. These had specially built tone controls which were effectively "out of the circuit" when the wipers were centered.

THE DIAGNOSIS

I hadn't played with one in many years, *until* one arrived the other day for "check out." Seems a young fella wanted to set up a basic tube system, picked up a PAS-3 for its preamp, and wanted me to look it over. Aware of the many mod choices available, he wanted to bypass the tone controls entirely. Now, I believe that a bass and treble control that is basically "out of the circuit" when in their mid-positions and which can add a "subtle" amount of boost or cut can be very valuable in a system. I asked whether I could leave them in but try to tame their effect.

Turns out that his PAS was in very good shape except for a reasonable amount of rust on the RCA connectors. Voltages were right on, and the filter caps were good. I then fired it up and used a great feature on my Amber Audio test set, which generates a continuously swept sine wave over the audio spectrum. The swept output emerging from the preamp can be monitored on an oscilloscope so you can view deviations from flat frequency response down to as little as a few tenths of a dB.

I first checked the line output stage by feeding the test signal into the "spare" input and verified that max bass boost/cut approached 20dB and treble 15dB. What surprised me, though, was that with the controls "centered," the flattest curve I could obtain showed a deviation of ±4dB (referenced to 1kHz) with the max plus deviation up around 8kHz or so and the max minus deviation around 200Hz.

THE MOD

What to do? Well, rather than "rip out" the controls, I spent a few hours with a bunch of resistors and clip leads. To shorten the story (a lot), I found a workable solution by simply paralleling both 750kΩ bass controls with 47kΩ fixed resistors (half-watt carbon) and the 400kΩ treble control with 33kΩ fixed resistors. Solder the lead from the side of each resistor to the "ends" of each potentiometer—the center terminal is the "wiper."

This resulted in a maximum bass and treble boost/cut of around 6 to 8dB and reduced the deviation from flat with the controls centered to ±2dB. Sweeping the phono section through an inverse RIAA network indicated that the frequency response was also within a few dB of flat—good enough for me.

Listening to the preamp brought back memories of why I liked my first one so much 40 years ago. In addition to its nice crisp sound, the bass and treble adjustment range was just about right. It was really quiet, too—after I cleaned all the connectors and potentiometers with Cramolin.

The only other mod I would suggest is if you plan on feeding an amplifier with an impedance less than 500kΩ, remove the 510kΩ output loading resistors soldered to the terminals on the back side of the RCA output jacks—just clip them out.

Hope the kid likes it. You might, too. *aX*

How Horns Work Revisited

This noted audio expert shares with us his experiences and insights into horn operation.

I read with considerable interest the recent series of articles on "How Horns Work" by Bjørn Kolbrek (March and April '08). Having spent the better part of my life studying these devices, I was anxious to read this discussion. I was somewhat disappointed to find so many errors in the discussion, particularly regarding my own work. I had considered responding to these errors individually, but then decided that this may not be the best approach. The discussion could become contentious, which might lead to arguments that were mostly semantic. What I have decided to do instead is to write how I believe horns and waveguides work, and the readers can sort out the errors for themselves.

Historically, horns served a very specific function, and it is important to understand this background because it highlights the motivation for the vast amount of work that I will attempt to shed new light on. That function was to improve the radiation efficiency of the audio playback system. (Please note that there is a distinct difference in the use and function of a musical instrument horn and the same device when used for music reproduction. I do not pretend to know what makes a good musical instrument horn, although I have read most of Arthur Benade's work, finding it, as expected, to be inapplicable to a reproduction horn. So please do not take anything that I say here as a condemnation or critique of the theory of musical instrument horns.)

Early playback systems had a serious problem generating sufficient SPL to be audible in the vast majority of situations. Efficiency was key to an effective design and the horn was well suited to this purpose. However, in a world where you have an almost unlimited availability of power to drive a reproduction device, you might want to question what use efficiency has in our current situation. Not that efficiency is unimportant or a bad thing, only that it is not really a major criterion in a modern design (there are exceptions such as in very large PA systems, but that is not what I am talking about here).

Efficiency is what Webster was concerned with, and nothing more. He wanted to know how much gain a device would have, and his approach works fairly well in that context. His work, however, has been extended well beyond the domain for which it is useful and into areas for which it is not even remotely applicable.

Webster assumed plane wave propagation in his analysis. Most people know this; however, very few ever stop to consider the implications. My favorite physics writer Phillip Morse (who will come up again) did discuss Webster's assumptions in his famous texts *Methods of Theoretical Physics*[1]. In these texts (two volumes) he defines precisely where Webster's assumptions are valid, namely:

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

where S_0 is the throat area and m is the flare rate. This is actually an extremely limiting restriction as I will show.

Consider **Figs. 1** and **2** (from *Audio Transducers*[2]). The **Fig. 1** plot shows the curve for an exponential horn $y_0 \cdot e^{x \frac{m}{2}}$ as an example, while the **Fig. 2** curve shows a plot of $\frac{d}{dx}\sqrt{S_0 \cdot e^{m \cdot x}}$. The region where Webster's assumptions are valid, according to Morse, lies where x < 15cm, or best case when x < 20cm (<< is usually interpreted to mean < .1). Beyond these lengths (x is the axis along the horns length), you can no longer rely on Webster's equation to be accurate. A line drawn from the center of the throat to its tangent point on the wall contour is shown in **Fig. 1**. This intersection happens at just about the point where the Webster's horn equation loses its validity according to the Morse criteria.

This coincidence highlights an interesting point, namely, that whenever the walls of the horn recede beyond the line where they can be illuminated by the center of the throat (as shown by the straight line), then the Horn equation fails to be reliable. This also happens to be exactly the same point where the wave-fronts would need to diffract if they are to remain in contact with the walls. It is exactly this inability of the horn equation to account for internal diffraction that is at the core of its problems. When there is no internal diffraction, then the horn equation will probably work well, but if there is diffraction, then you must seriously question Webster's approach.

How is it, then, that the horn equation has apparently worked so well in the field for all these years? For "bulk" quantities, such as the input impedance, it does work well. This is because the acoustic impedance is an average of the wave-front details across the waveguide. As long as you are not concerned with these detailed motions, such as in an impedance calculation, then Webster's equation works satisfactorily.

The Horn Equation does an admirable job of predicting the impedance of a wide variety of horn contours, which is completely in line with Webster's goal of finding the "gain" of the device. And it does this through a fairly simple calculation procedure. What's not to like? But the price that you must pay for this simplicity is a complete lack of knowledge of the wave-fronts details across the contour—the horn equation is blind to these details.

DIRECTIVITY

For a variety of reasons, directivity control is a highly desirable feature in loudspeaker systems, and such control is basically unavailable without the use of a waveguide. Much work has been done on directivity control in a waveguide, but much of it is misguided.

You can draw a complex set of wave-fronts on a piece of paper and connect them with a contour, but that does not mean that real sound waves would necessarily follow these drawings. Sound waves have a mind of their own and they *will* go wherever they want, no matter how much hand waving you do. Take, for example, defining horn shapes based on the assumption that the wave-fronts propagate with a constant radius.

This assumption yields some very nice looking curves and horns. Unfortunately, it is physically impossible for a wave-front to propagate in such a manner—it violates every concept in physics from Huygen's Principle to Green's Theorem. The wave-fronts in the real device *will not* follow the wave-fronts as drawn on the paper. It's just not that simple.

Furthermore, a horn contour that is based on a geometrical "plot" of how the wave-fronts will progress relies on faith that the wave-fronts will actually propagate that way. This naïve faith is no way to proceed in a scientific study of a real device. You have no option but to search for a mathematical description that defines precisely *how* the wave-fronts actually will propagate using physics, not naïvely prescribing a situation that you *hope* will occur.

To predict or design for directivity, you must know exactly what the wave-fronts look like at the mouth of the waveguide, and this is not so easy—certainly not as easy as just calculating the impedance. This is precisely the point where I found myself in the 80s. I wanted to do accurate directivity control in a horn, but I realized that none of the mathematical descriptions available at that time could achieve this.

I had studied the early work of Morse in *Vibration and Sound*[3] and found his concept of 1P (for one parameter) waves compelling. It was natural that I looked for solutions along these lines. Making a long story short, to be 1P means that the waves must be described by a coordinate system of the so-called orthogonal variety.

Orthogonal coordinates are well known, and Morse's *Methods. . .* [1] devotes an entire chapter to this topic (chapter V, my favorite, one of the most elegant discussions in all of my physics literature). It can be shown that you can define exactly 13 (and only 13) coordinate systems that are orthogonal in three dimensions. However, of these 13, only 11 allow solutions to the wave equation. In other words, sound waves can be described mathematically by only 11 coordinate systems.

It became readily apparent to me that only in these 11 coordinate systems could you ever hope to find a mathematical description along the lines of the 1P idea. The problem is that of these 11 only two actually allow for true 1P behavior. The others, while still orthogonal, are dimensionally coupled spectrally through their so-called separation constants and end up not being 1P. In essence, this means that the wavefronts "leak" energy from one coordinate into the others—basically the diffraction aspect that I mentioned earlier.

THE COUP DE GRÂCE FOR 1P

It is interesting to note that the two (reasonable) 1P possibilities are the radial coordinate in spherical coordinates and the radial coordinate in cylindrical coordinates. The two horns that are defined by these coordinates turn out to be exactly the same two horns that Putland discusses in his paper on Webster's equation[4]—which deserves a short diversion. "Didn't Putland show that any solution of Webster's equation is a 1P solution?" Yes, that's absolutely correct, and he also proved that Webster's equation is exactly correct for only two horns, not coincidentally the same two that I defined above through a completely different line of reasoning.

I had actually arrived at this exact same conclusion some years before Putland (in *Acoustic Waveguide Theory*[5]), but I guess he must have missed it because he didn't mention it. The claim made by Mr. Kolbrek that "Putland later showed that this was not the case," is incorrect because he did no such thing. Putland also did not indicate a recognition of the existence of the higher order solutions (they aren't predicted by Webster's solutions, but do exist in the wave equation approach—it is these details that make all the difference!).

You see, in order for a 1P wave to propagate within a 1P device, it must be excited by a purely 1P source wave-front. Any other wave-front will excite higher order modes (HOMs), and the wave propagation will no longer be 1P. In the case of a conical horn (from spherical coordinates), that's a spherical cap that vibrates radially (normal to the curved surface). Unfortunately, such a source device does not actually exist; you could hypothesize making one, but they don't actually exist today.

A dome loudspeaker vibrates axially, not radially, and so it is not 1P. A conical horn on a dome source is not 1P. For the horn in cylindrical coordinates the source must be a radially vibrating cylindrical section, which, once again, does not exist in practice.

So while Putland may have been correct in his *J*AES article[6], this fact is of no practical use because 1P devices can only exist in a hypothetical world, and real-world devices are never 1P. It is no coincidence that both Morse and I ceased to use the concept of 1P after studying the problem because we both realized that such an ideal solution was only of academic interest in a hypothetical situation.

What I have said thus far:

• Unless you are only interested in the impedance of a device, Webster's horn equa-

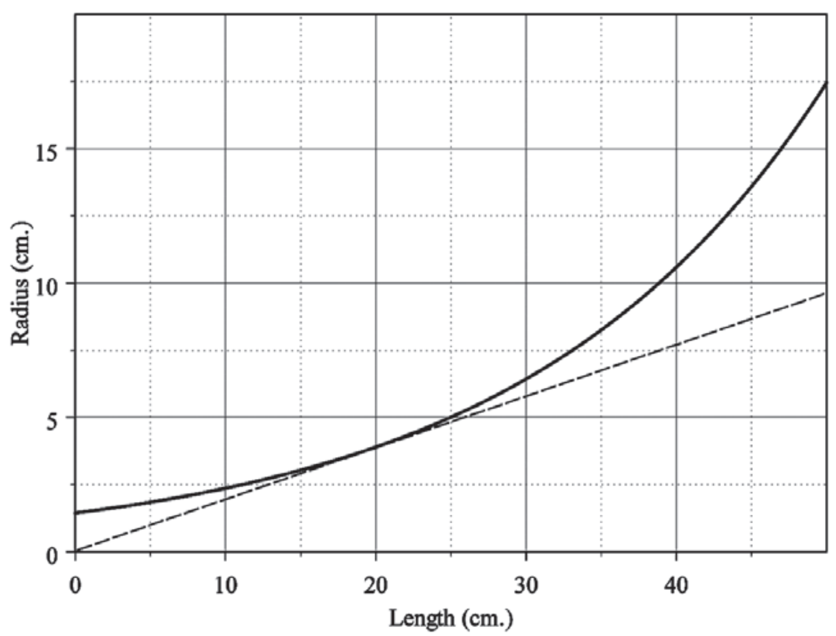

Figure 1: Typical exponential horn contour.

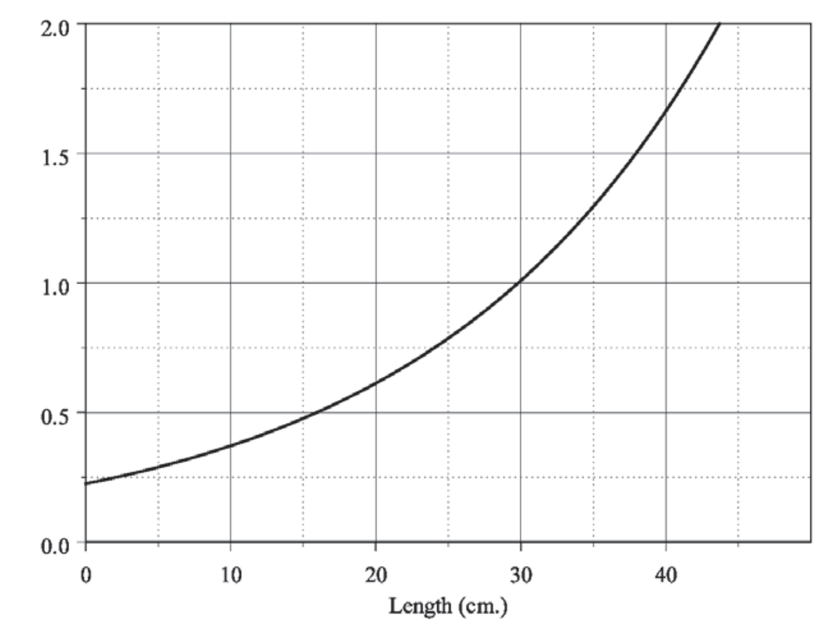

Figure 2: Plot of equation.

tion is not of much use, and even then it works correctly only in a hypothetical situation of no practical interest. To me, any proposition that uses the horn equation at its foundation needs to be viewed with extreme skepticism.

IMPEDANCE (AGAIN)

"Well, at least the horn equation is good for predicting the shapes that one needs in order to achieve good impedance loading—at least at low frequencies!" Sure, except that it turns out that virtually any shape connecting a given throat and mouth area will yield approximately the same impedance—within a dB or so. In the long wavelength region where the horn equation is valid, shape is simply not a significant factor; the waves don't see shape. Basically, they only see the inlet and the outlet areas and the distance between them and everything else is of secondary importance. But, if something secondary is important, then it is almost certain that Webster's equation won't apply.

THE EXACT SOLUTION OF THE WAVEGUIDE EQUATIONS

Contrary to some widely propagated errors (one of which is in Mr. Kolbrek's article), I actually did develop an exact solution for a waveguide using the Oblate Spheroidal (OS) coordinate system[5, 6]. It was in these AES papers that I came to realize the importance of HOMs, which the horn equation fails to account for and which make true 1P behavior unrealistic. All horns and waveguides in the real world have HOMs, which produce the complex wave motion that prevents us from prescribing it. Whenever we attempt to force the wave-front to travel a path that it doesn't want to, it gets back at us by generating HOMs.

The advantage of waveguides based on the orthogonal coordinate geometries is that it can be proven that they will generate the least HOMs of any contour connecting the same input and output apertures. No shape can do better. That, to me, is the attraction to the approach and the OS waveguide for a standard circular source.

It is interesting to note that if you draw the OS contour as in **Fig. 1**, that a line from the origin to the walls is never tangent to the walls as it is in the exponential (and virtually every other) horn.

I have solved the waveguide equations in just about all of the geometries for which the theory applies and designed and built waveguides in them. They all work well, but with different constraints on the polar patterns, source configurations, and so on. Different coordinate systems can also be combined such as in my *Bi-Spheroidal™* waveguide (patent #7,068,805) in order to achieve specific objectives.

Readers interested in more detail on waveguide theory can find them in "Chapter 6—Waveguides" of my book *Audio Transducers*[2](available through Old Colony).

I would like to point out why I coined the term "waveguide." It was done to highlight the fact that the horn equation was not used and as such a new name was appropriate. Unfortunately, usage of this term has gotten out of control and it is now applied to whatever anyone wants to apply it to.

With a waveguide, a complete knowledge of the wave-front is available at any point, the most useful surfaces being the mouth and the throat apertures—the interfaces. If you have a desired directivity pattern, then the wave-front shape required in the mouth aperture can be calculated—assuming that such a pattern is obtainable within the given mouth size. The calculations will, however, define what the mouth size needs to be in the event that it is too small or what must be given up for a fixed mouth size. With this knowledge of the mouth wave-front, you can then calculate the wave-front required at the throat. Now, in principle, you could fabricate a phase plug that achieves the required throat wave-front (it's not likely to be a plane wave). Because the wave shape may need to change with frequency in order to achieve the desired polar pattern, there is no guarantee that such a solution is possible or practical. But it is guaranteed that no other design could do any better because with waveguide theory you can get right up to the limits of what the physics of the situation will allow. Such a detailed design is simply not possible with horn theory.

DIFFRACTION

Diffraction—that aspect of horns that horn theory cannot deal with—turns out to be of predominate importance to sound quality. Diffraction in a waveguide is a bad thing, a very bad thing as it turns out. It is not readily masked in the ear, and any masking that does exist decreases with sound level[7], meaning that diffraction becomes more audible at higher sound levels—the exact opposite of nonlinear distortion. Diffraction is what you hear in a waveguide as harshness or poor sound quality. Reduce the diffraction and the device simply sounds better.

Diffraction can occur in a number of places including the phase plug. When the waveguide is not presented with the proper wave-front as prescribed by its specific geometry, then a form of diffraction occurs in that HOMs are generated. Any form of discontinuity anywhere in the device can produce this problem.

In my experience no amount of attention to the detail of the throat geometry and its wave-front is unwarranted. Even small discontinuities in the throat matching or errors in the phase plug design will generate diffraction in the form of HOMs, which then become amplified by the waveguide. Specific attention to the phase plug is of paramount importance (patent # 7,095,868 and patents pending).

THE PIÈCE DE RÉSISTANCE

Only quite recently I had an epiphany regarding waveguides versus horns. Horn theory is concerned only with the rate of change of area as it relates to the impedance presented by the device. No consideration is actually given to the shape of the boundary beyond the effect that it has on the area. Waveguide theory on the other hand is the exact opposite in that no explicit consideration is given to the area—it is what it is—and only the shape of the boundary matters.

The key point is that all of the 11 orthogonal coordinate systems (those that allow for expanding waveguides as opposed to parabolic ones) have one very important characteristic in common: every one of them has exactly the same shape boundary, either straight or of the form

$$y(x)\sqrt{y_0^2 + \left(x\,\mathrm{Tan}(\theta_0)\right)^2}$$

where y_0 is the throat radius and θ_0 the wall angle. This curve is known as a catenoid, the minimum length of a curve joining two points with prescribed angles at each end point, with the one at y_0 being zero (modifications for non-zero entry angles are possible). A straight line is the catenoid when the two points have the same angles.

In short, horn theory is concerned with the rate of change of area and impedance while waveguide theory is concerned with the rate of change of the boundary and the ensuing wave-shape. Waveguide theory seeks to minimize the generation of diffraction within the device but Horn theory, unfortunately, does not even consider it.

CONTROLLING HOMs

Having realized the importance of the HOMs, you can do several things that are quite separate from the waveguide design to minimize these undesirable characteristics. I have learned that the inclusion of open cell foam within the body of the waveguide will reduce the HOMs much more than the primary wave, thus yielding a pronounced net benefit in sound quality (patent pending).

MY POV

After 30 years of study, I have come to several conclusions that I will highlight here:

1. The horn equation is of no practical use in any useful device.
2. Impedance calculations are of marginal utility—I never do them for my designs—and the concepts of loading, cutoff, and so on are all obsolete holdovers from horn theory.
3. Nonlinear distortion in a waveguide is irrelevant. This comes from the fact that this distortion is of very low order and as such is inaudible[8,9].
4. Waveguides are primarily useful for directivity control and are the only practical way to achieve this critical design objective.
5. Sound quality goes in inverse relation to the presence of HOMs, and attention to detail in this regard is essential.
6. Waveguides are not very useful for low-frequency transducers. The directivity control function of the waveguide is defeated when its size is small compared to the wavelength; efficiency and low distortion are not that important at these frequencies; and I can get better efficiency and lower distortion using a more efficient approach such as an acoustic lever (patent # 6,782,112) or a bandpass design[10]. There are some situations where a horn's good efficiency over a larger bandwidth than levers or bandpass designs can be useful, but there are also serious trade-offs both ways. I have not found horns to be cost and space efficient at low frequencies in any of my work.

There will be a multitude of people who will disagree with these conclusions and they will use all kinds of arguments about what "sounds better." That's fine, they have every right to their subjective opinions. But mathematics is not subject to opinion nor open to debate, except with alternative *mathematical* arguments. Science is not a democracy that can be voted on with the popular opinion becoming reality (except for evolution).

The reader will note many areas where I disagree with Mr. Kolbrek, but mostly I disagree with continuing to propagate concepts that are not based on reliable science, having been derived from a misguided application of horn theory beyond its very limited applicability.

I would like to thank Mr. Kolbrek for his good historical discussion and many good references, some of which were previously unknown to me. I hope that he will find my arguments compelling enough to embrace waveguide theory as the only viable approach to the problems at hand.

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Bjørn Kolbrek responds:

I would like to thank Dr. Geddes for his interest in my article, and for his comments. He raises many valid points. I will attempt to address the key ones.

I should perhaps begin by reiterating the intent of my article, which is clearly outlined in the title ("Horn Theory: An Introduction") and the opening paragraph: The article ". . . reviews the basic assumptions behind classical horn theory as it stands, presents the different types of horns, and discusses their properties." It is therefore unreasonable to expect it to give a complete account of acoustical waveguide theory.

That the Webster horn equation is nothing more than an approximation was pointed out in my article, and as an approximation it can only deal with certain aspects of what it describes. Many of the limitations of the horn equation have been known since the 1920s, as a reading of the references I have quoted will show. As Dr. Geddes says, Webster's horn equation deals with acoustical impedance. I have never claimed otherwise. I do not, however, believe that the application of the horn equation is as limited as Dr. Geddes says. It is true that it is only completely valid within a small range. But the limits of approximate validity can be stretched quite a bit, and several investigators have shown that the throat impedance of a horn can be calculated with good accuracy (compared to measured results), if you abandon the plane wave assumption and make use of a curved isophase wave-front area expansion in the calculations.

However, if you want to calculate directivity and higher order effects, then I agree that the horn equation is not applicable. I believe I made that point clear in my article. I also tried to show that horn design is not as simple as deriving the profile from an assumed set of wave-fronts. Sound waves will do "whatever they want," and I believe Figs. 15 and 16 from part one of my article show this clearly enough.

Dr. Geddes has rejected the horn equation as having no practical use. From that point of view, every treatment of horn theory based on this equation will by definition be looked upon as being "full of errors." I believe that many of the errors he has found in my article come from the fact that I did not reject the horn equation completely.

That I have made factual errors is clear. I have obviously misunderstood the Geddes/Putland debate and some of Dr. Geddes' work. It was perhaps also a mistake to include the OS waveguide in a discussion on classical horn theory, because this device is best analyzed using waveguide theory. For these errors I offer Dr. Geddes my sincere apologies.

That said, I don't understand how Dr. Geddes can claim that all horns that have the same end areas and length will have virtually the same loading characteristics regardless of profile, without giving the conditions under which the assertion holds. It is probably applicable to short horns, in the range where the horn equation is valid. If it is believed to be true for a larger range of horn profiles and sizes, then I would welcome seeing the practical data, because all the published measurements that I am aware of show the opposite. It is true that all horns approach the asymptotic throat acoustical impedance of

$$Z_A = \frac{\rho_0 c}{S_t}$$

but the frequency at which this occurs is not the same for all horn profiles, no matter how much "hand-waving" is done.

When throat impedance (and resistive loading of the driver), distortion, and efficiency are all rejected as unimportant, then classical horn theory does not have much to offer. Designers concerned only about directivity control and higher order effects are better off with waveguide theory. It is, however, my view that classical horn theory still has its uses, but you must keep in mind that you are working with approximations.

 By Ed Simon

2400W of Clean AC Power

The fifth in this series of articles on cleaning up AC power lines.

In the first four parts of this series, I explored ways to provide superior AC power for audio systems. Here I look at expanding the system to provide 2400W of clean power. You do this by combining the balanced isolation transformer and oscillator-amplifier combination (**Fig. 1**).

The trick is that you need a clean reference oscillator tied to the AC line. You could use the previous transversal oscillator and phase-lock the 555 timer circuit. You would still need a lockout for when it is not in sync. This approach is quite doable. I, of course, chose a different, older, approach.

OSCILLATOR CIRCUIT

The almost forgotten circuit I prefer is the synchronous oscillator. This is a devious leftover. In any oscillator there is an amp with a frequency-selective level-dependent feedback. When the feed-forward and feedback have 360° of phase shift, it oscillates if the gain is greater than the loss of the feedback network. The oscillation will continue to grow in amplitude until the amplifier clips, which is a very high distortion oscillator. That is why a monitor circuit is used to reduce the gain until the gain exactly matches the feedback loss. This is, of course, impossible to do perfectly, so all oscillators must have some minute residue of distortion.

In the synchronous oscillator I cheat! The gain is just slightly less than the loss through the feedback network. I then inject some of the desired signal into the feed-forward feedback path. This starts the ball rolling and keeps it going. If you limit the input voltage, you achieve a stable, cleaner synchronized output. You can now use a crude limiter on the input signal to keep it stable. A simple circuit, with few adjustments, gives good results.

In **Fig. 2** J1 gets 24V AC from a center-tapped transformer. I tapped this off the existing amplifier transformer. The four diodes rectify this and provide ±17V (usually higher due to the light load on the transformer) to the 220µF filter capacitors. The regulator chips turn this into ±12V. The output capacitors are small enough so that when you turn off the AC input the regulators will not be reverse-biased. The current draw of the circuit is only a few milliamps, so no heatsink is required for the regulators.

J2 is designed to accept the reference signal. This can be from extra turns of wire looped around the transformer's core, the 24V transformer through an additional 47K resistor, or anywhere else you can get a sample of the AC line. The input voltage here should be a volt or two. For my version the reference input is a few turns wound around one of the amp's power transformers. It gives a good phase match to the output windings.

ADJUSTMENTS

R1 is the input level adjustment. You should set this so that the output is into clipping. The resistors R2 and R3 set the gain of the amp at 11. Diode D5 with the diode bridge provides complete feedback if the output exceeds 6.2V. Two resistors bias D5 to keep it always on, resulting in cleaner clipping action. This first half of U3 is a classic voltage

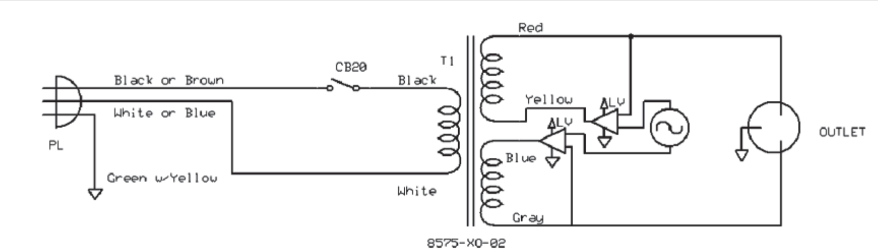

FIGURE 1: Center-tapped transformer circuit.

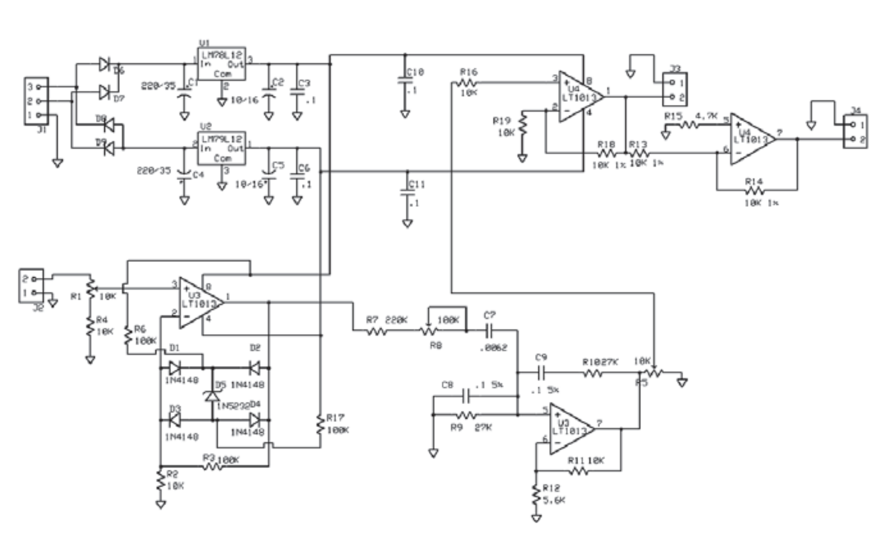

FIGURE 2: Oscillator circuit.

limiter amp.

If you use an oscilloscope or equivalent, adjust R1 to get a flat top but not a square wave. If you have just a voltmeter, adjust R1 until the voltage slows, rising as you turn the adjustment pot, then give it a bit more. I ended up setting it for the lowest consistent voltage, because that gave me the best final output regulation.

The output at pin 1 U3 still will change a bit with large swings in the AC synchronizing input voltage, but it seems to be fine for my use. This voltage feeds the other half of U3 through another adjustment, R8. This second half of U3 is a classic Wien bridge oscillator, except the gain of R11 and R12 is set too low for it to oscillate by itself.

R8 inserts the clipped signal into the feedback loop and is used to adjust the precise oscillation phase. This is the fine phase adjustment to make sure your zero crossings from the reference output and the AC line are the same. This made no difference in the output voltage or the output impedance, but it should affect how much heat the correction amps generate. I left it at midscale. With a 1500W load, my heatsinks hit 57° C, so I left it alone. You may wish to try different settings to see how you get the least amount of heat.

The correction amps become the hottest when they are doing the least work! If there is no correction voltage being applied, the output of the correction amp will be at zero volts. However, all of the current that goes in or out of the AC cleaner must also pass through the correction amps. That current multiplied by the rail voltage is the power the amp must dissipate! When there is a correction voltage, there is less voltage drop across the output transistors, resulting in less heat. This is quite different from most amplifier applications, which is why I used such a big heatsink.

The final voltage adjustment is R9, which sets the level going to the correction amps. You should set it so that the final output voltage is 120V RMS or PRRI. Both readings should be the same if the unit is working.

I also added U4, which is a buffer to make sure U3 is not loaded and more stable. It also provides a 180° phase shifted output. To divide the work using two smaller amps, one pushes while the other pulls. I chose the LT1013 for low power consumption, reasonable drive level, and low DC offset.

The amplifier for this is required to handle a lot of current. If you use the 20A transformer from Plitron, you should expect 20A through the amp circuit. Just my luck (or maybe I already had this goal in mind), the amps I designed for the smaller power supply will work just fine!

I modified the 300W cleaner circuit by replacing the output transformer and changing the oscillator card. You may also wish to beef up your PC card by soldering 12 gauge wire to the entire set of high current traces. If your PC board has thin copper foil, it could vaporize, or at the least add resistance where it hurts.

POWERING UP

I now had a device that was about to handle 2400W or much more if there was something wrong (**Photo 1**). Plugging it in with some errors would instantly turn a lot of work and more than small change into smoke or flames. I decided to go slow.

The first time I turned it on through my light bulb box, I connected only one amp to one of the Plitron isolation transformer's secondaries. I made sure I had the phasing right. When it was not, I just changed the feed from the oscillator to the other reverse phase output. I duplicated this for the other side. Once both sides worked, I connected one output lead to the hot on the AC receptacles, the other went to where the neutral would normally go. The amp's common ground is the center tap of the balanced supply and was bonded to the outlet's safety ground.

Everything seemed great. I tried a small load and watched as after a few seconds the output voltage dropped. I had forgotten that the light bulb box limited my current draw.

I plugged the AC cleaner into the outlet directly. It blew the outlet's circuit breaker. The isolation transformer is good for 20A and the correction amps each use another 2 or 3A. The outlet my AC cleaner was plugged into was only rated for 15A.

PHOTO 1: Inside the cleaner unit.

I reset the circuit breaker but that wasn't enough. A bit of troubleshooting showed the actual outlet I had plugged into failed! I changed the receptacle and decided it may be wise to keep the 2Ω 15 to 25W slow-start circuit for this version also. If you ever want to use this supply for the full 20A capability, you will need to source it from a 30A circuit.

The unit ran dead cold for small loads. The improvement in efficiency over the 300W version was quite noticeable.

I put on a 1500W load and watched the voltage drop by less than 4V. This gave me an equivalent source impedance of about 300mΩ. I could try turning up the correction amp gain if I wanted a stiffer supply, but that seemed a good place to start. Too stiff a supply may cause rectifier failures in the gear it is powering.

There are several additions you could make to this or the other power line cleaners. The next improvement would be to have a sequencer automatically cycle the equipment on and then reverse order to turn it off. Adding a soft-start to the sequencer would also be a nice idea. I figure that if you want to build one of these, you should be able to do that part yourself. Of course, if you do, you might want to let others know about it.

One problem that this and the 300W supply have is that they also draw current for the amps from the peak of the AC voltage. Thus they are dirtying their own source of power. A different type of power supply would avoid that problem, but that is a different article.

But, in audio, there is always a better way. A simpler lower-cost approach to the problem would have no active electronics. It would have to store the energy in a simple manner and be able to return it when needed.

OTHER OPTIONS

There is just such a device that has been around for a long time, older, even, than the vacuum tube. It is sometimes called a cycloverter or motor generator. I can connect a motor to a generator that directly produces AC (technically an alternator).

I had a 1000W rated gasoline powered generator (**Photo 2**); looking under the hood showed it would be difficult but not impossible to hook it up to a 3600 RPM two horsepower motor. That would give me a motor generator set for less than the cost of the isolation transformer if I bought well at the junkyard!

Using the generator with the internal gasoline motor showed that the output varied a bit with the load both in level and frequency. This is not very green and not a very good approach.

There are complete commercial units designed to convert single-phase 220V AC service to 230V three-phase. The smaller units are just capacitor based, but the larger units are motorized. One of these could be the basis of an off-the-shelf solution. However, the hidden components in an AC generator or alternator cause distortion. So you may still want to use a cleaner such as the resonant filter. For a very complete discussion of this see Chuck Hansen's articles in *aX* 9/01 and 10/01.

Of course, you should not rule out the other options of a wind-, solar-, or water-powered generator. Making your own clean green power might not be a bad idea.

If you are tempted to play with one of these approaches to cleaning your power line, you may wonder how these affect the quality of the reproduction of music?

Bigger and more expensive does not need to correlate with quality. The simple voltage stepup transformer is a good choice if you are listening with loudspeakers of limited efficiency. You can use the unit as a stepdown with older classic gear to get it back to the design voltage.

The resonant filter is probably the best cleaner for the buck, but it does make some acoustic noise. I also do not like that the inductors are not rated for this use. I also found that some gear does not sound as good with a pure sine wave. It seems the third harmonic distortion adds more charging time to the power supply and improves sound quality!

The isolation transformer boosts the voltage and cleans up the line. This improves sources and preamps as well as the power amp.

The complete regeneration of the AC line as in the 300W version is probably the least cost-efficient and least green version of all the cleaners. To get around this problem I am currently setting up a windmill to use the 300W version to power my class A system. Almost an eco-friendly compromise!

The corrected AC regulator is probably the best at doing the job. Keep in mind the one I built is able to provide 20A. This same design can be made much more economical if you design one for 600 to 750W.

The motor generator is the most powerful of the approaches because it is the simplest and can be combined with other energy input sources. It is not the best sounding from my simple trial. The best solution is a well-designed power supply in the audio gear itself. *aX*

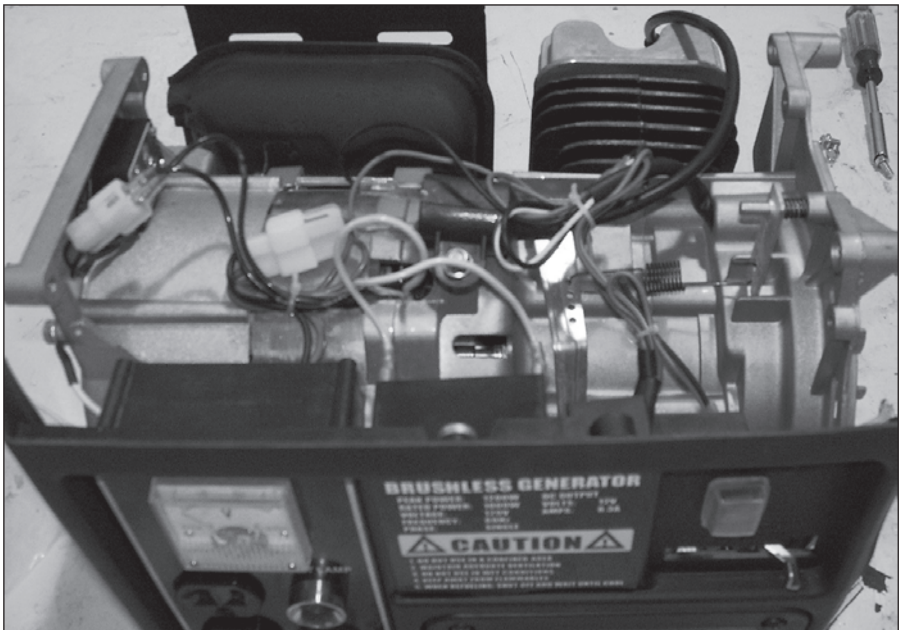

PHOTO 2: Gas-powered AC generator.

Tascam CD-RW900SL CD Recorder

By Gary Galo, Regular Contributor

Teac Corporation
7733 Telegraph Road
Montebello, CA 90640
323-726-0303
www.tascam.com
$699
12.2" × 19" × 3.7"
(309mm × 483mm × 94mm)
10.4 lbs. (4.7kg)

PHOTO 1: Front view of the Tascam CD-RW900SL CD recorder. Rack mounts are included with the recorder.

The CD-RW900SL is one of two stand-alone, single-disc CD recorders manufactured by Tascam, the professional audio division of Teac. The CD-RW900SL is an updated version of the CD-RW900, which improves on its predecessor mainly in added features. As Tascam notes in its product overview, "Recently added features include Auto Cue and Auto Ready from the menu, selectable I/R remote Enable/Disable, and MP3 Action Setting (prevents accidental termination of continuous MP3 playback)." The earlier version also had conventional tray loading; the "SL" version has a trayless, slot-loading transport. The "SL" version also has 24-bit A/D and D/A converters (the previous version may have, but the manufacturer didn't specify).

Although the list price of the CD-RW900SL is $699, street prices vary and are often considerably less. I bought mine on eBay for $439, but there's one catch: Make certain that you buy from an authorized Tascam dealer, and that the unit you intend to purchase is covered by the US warranty. My eBay dealer was Jim's Music Center, which met both criteria. They have a 99.9% positive rating, shipped the same day I ordered (and emailed me the UPS tracking number), and provided a one-year extended warranty for free. As of this writing, a number of dealers are still selling the original version; if you want the new features, make sure yours has the "SL" suffix.

ANALOG AND DIGITAL I/O

The CD-RW900SL has unbalanced (RCA) analog inputs and outputs, plus S/PDIF coaxial (RCA) and Toslink optical digital inputs and outputs. Tascam also makes the CD-RW901SL, which adds balanced XLR analog inputs and outputs, plus AES/EBU digital I/O. The 901 version also allows external control via both 9-pin RS-232C serial and 15-pin parallel "D" connectors, and is supplied with a *wired* remote control (not infrared) that you must plug into a mini-phone jack on the back of the unit. Otherwise, the two versions are identical (if you prefer a wireless remote, you'll want the unbalanced 900 version, reviewed here).

The Tascam records on standard CD-R and CD-RW (rewritable) discs. Because the unit is a professional recorder, you can use any recordable CD media—data or music. Even if you are not a professional user, I would consider purchasing a pro CD recorder rather than a consumer unit. With a consumer unit, you must use "music CD-R" media, which are often twice the price of regular CD-R media. If you do a lot of recording, you'll save money using a professional recorder (I am assuming that the recorder will be used for legal purposes only; don't use any recorder to violate copyright laws!).

Tascam's technical sheet on the CD-RW900SL lists approved brands of CD-R and CD-RW media. In my work as audio engineer at The Crane School of Music, SUNY Potsdam, I use mostly Mitsui Gold CD-R discs for long-term archiving and Fuji discs for general-purpose distribution. Neither is listed among the approved brands, but both seem to work fine.

The safest bet is to stick with reputable, name-brand discs, and steer clear of the cheap no-name or "house brand" discs sold by your local office supply store. With CD recording, there's no substitute for reliable and repeatable performance, and with blank media you get what you pay for. The Tascam will work with media designed for high-speed duplicating, so you don't need to search for discs designed specifically for "1×" (real-time) recording.

PLAYBACK FUNCTIONS

The Tascam recorder features a fluorescent display that offers a generous amount of data. You can activate most of the CD-RW900SL's functions, including the menu system, from the front panel. The wireless remote control makes getting to some of them easier, however.

The Tascam remote has an intelligent keypad system for track selection. Only "0" through "9" appear—there is no "10" button for selection of tracks 10 and higher. Simply enter the track numbers directly—2 then 3 for track 23, for example—then press the play button. The "direct search" function allows you to go to a specific place within a track. If you want track 3 at 1:15, simply enter 003 001 15 on the remote, and the player will begin at that spot.

With many CD players, you must toggle the skip button to select a specific track if you're not using the remote. The CD-RW900SL has a jog wheel—which you can also use for menu selection—that allows you to simply dial up the track you want (then press play). The time key on the front panel or remote allows you to toggle between elapsed track time, remaining track time, total elapsed disc time, and total remaining disc time.

The Tascam supports other playback functions that have become standard on most CD players, including random playback, repeat playback, and programmed playback. "A-B" repeat playback allows you to select start and stop points within a track (but not across tracks). Auto-cue allows playback of any track to commence after the level rises above a predetermined point, and audio-ready cues up the next track after a single track has been played. A timer playback function allows you to start a previously loaded disc when the player is powered up from an external AC timer.

The Tascam will also play MP3 files, and the "directory" mode allows you to select nested directories and play/select files within each directory. CDs that contain MP3 files can be played on the CD-RW900SL as long as there are no more than 999 MP3 files on one CD. The track display on the recorder is three digits long, but the Red Book CD specification limits the number of tracks on a CD to 99. The three-digit display is only used to select MP3 files above 99, or when using the direct search feature on CDs, as noted.

PROBLEMATIC PITCH CONTROL

Tascam has made several CD players over the past few years with variable pitch. My main interest in variable-pitch playback is to correct commercial CD reissues of historical material when the original recordings have been transferred at the wrong speeds. Unfortunately, Tascam's previous variable-pitch players have allowed adjustment in only 1% increments, which is *way* too coarse. They must have responded to complaints about their pitch controls: the CD-RW900SL has a pitch control which operates up to ±16% in 0.1% increments (you can change the increment to 1.0%, if you wish). Most variable-pitch players will allow adjustment up to ±12%, which is a whole tone, but the wider range of the Tascam will be welcome for severe errors on the part of transfer engineers (I've seen recordings off by a minor 3rd, which even the Tascam won't quite fix).

Generally, if a CD player has a pitch control, the sample rate at the digital output will change as the pitch is changed. This will cause many outboard D/A converters to unlock, as will some other digital recorders, unless a sample rate converter is inserted between the player's digital output and the digital input of the device being fed. The Tascam CD-RW900SL has a unique feature—the built-in sample rate converter also functions in playback, keeping the digital output

PHOTO 2: Rear view of the Tascam CD recorder. The CD-RW900SL has unbalanced analog inputs and outputs, plus S/PDIF and Toslink Optical digital I/O. The CD-RW901SL version adds balanced analog connection, plus AES/EBU digital I/O.

at 44.1kHz when the pitch control is activated (when the pitch control is off, the sample rate converter is not activated). This would be an extremely thoughtful feature if it had been implemented properly, since it would eliminate the need for an outboard sample rate converter.

Unfortunately, as Chuck Hansen notes in his measurements (see accompanying article), the high frequency response rolls off when the pitch control is engaged. This happens even when the player's digital output feeds an outboard D/A converter. I have three other variable-pitch players and none has this problem (Denon DCD-1015, Denon DN-600F, and Marantz PDM-340). Chuck also notes an increase in distortion when the pitch control is turned on, even if the pitch is set to 0.0%, which is another annoyance.

The high THD Chuck observed with the pitch control engaged is also unacceptable. If the Tascam is turned off with the pitch control engaged, it remains that way the next time it's turned on. My Marantz and Denon players all default to "Pitch Off" when powered up, which is much more sensible.

The digital pitch control functions just like an analog pitch control, changing both pitch and tempo. This is true whether you're monitoring the unit via its analog or digital outputs. If you wish to change the pitch while leaving the tempo unaltered, Tascam has also included a "key" control, which allows you to change the key up to ± half an octave, in half-tone increments. If the key and pitch controls are activated simultaneously, the pitch control will change the playback speed while leaving the pitch (key) unchanged. DJs may find the key control useful. The pitch and key controls will not operate on MP3 files.

RECORDING DETAILS

Basic recording is extremely easy on the CD-RW900SL, and can be done via the analog or digital inputs. Like most modern CD recorders, the Tascam allows you to adjust the record level for both analog and digital sources. You can also monitor recording levels even if a disc is not inserted by simply pressing the record button. The word "Monitor" appears in the display, and the recorder functions as a D/A converter, with the fluorescent meters showing record levels.

Most CD recorders have come with a built-in sample rate converter on the digital inputs, including the ancient HHB CD-R800 (my first CD recorder). When I was recording concerts to DAT, I always recorded at 48kHz, and then made copies to the HHB, which provided the necessary sample rate conversion automatically. The Tascam CD-RW900SL also has a built-in sample rate converter, but you can turn it on or off in the record menu.

If the digital source is within 1% of 44.1kHz, the recorder will lock onto the incoming signal. If it's off by more than 1%, you'll get the error message **Not Fs44.1kHz!**, in which case you should activate the sample rate converter. If the source is 44.1kHz, it's better to leave it off; otherwise, the incoming data will be re-written even though it doesn't need to be.

The CD-RW900SL has a number of advanced recording features, including automatic fade-ins and fade-outs. The fade-in or fade-out functions must be activated with the remote control. You can sync-record with both analog and digital inputs. With analog inputs, the recorder can add a new track each time the level rises above a predetermined level, which you can adjust using the sync level option in the record sub-menu.

In the digital direct (DD) sync mode, a new track will be entered each time one is detected in the source. This is extremely useful when making direct digital copies of CDs, DAT tapes, or MiniDiscs. You can also manually add tracks by simply pressing the record button while the unit is recording (each track must be at least four seconds long, as dictated by the Red Book specification).

When you are finished recording, you must finalize the CD. This writes a table of contents, making the disc playable on a regular CD player. The Tascam recorder finalizes a CD in about 100 seconds, the fastest I've seen in a stand-alone recorder. My old HHB CD-R800 took over four minutes; the more recent HHB CDR-830 and the Sony CD-RW66 will do it in just over two minutes.

Once a CD is finalized, you can't add any more material to the CD. The Tascam recorder allows you to "unfinalize" a CD-RW, which erases the table of contents, allowing you to continue recording on the disc. On CD-RW discs, you can also erase either the entire disc or individual tracks.

The Tascam also allows you to add titles to discs and individual tracks, using the text edit function on the text submenu. Entering text is a bit cumbersome unless you have an external keyboard. Fortunately, the CD-RW900SL has a PS/2-style keyboard jack on the front panel, allowing you to plug in any computer keyboard with a PS/2 connector.

Since bailing out of the defunct DAT

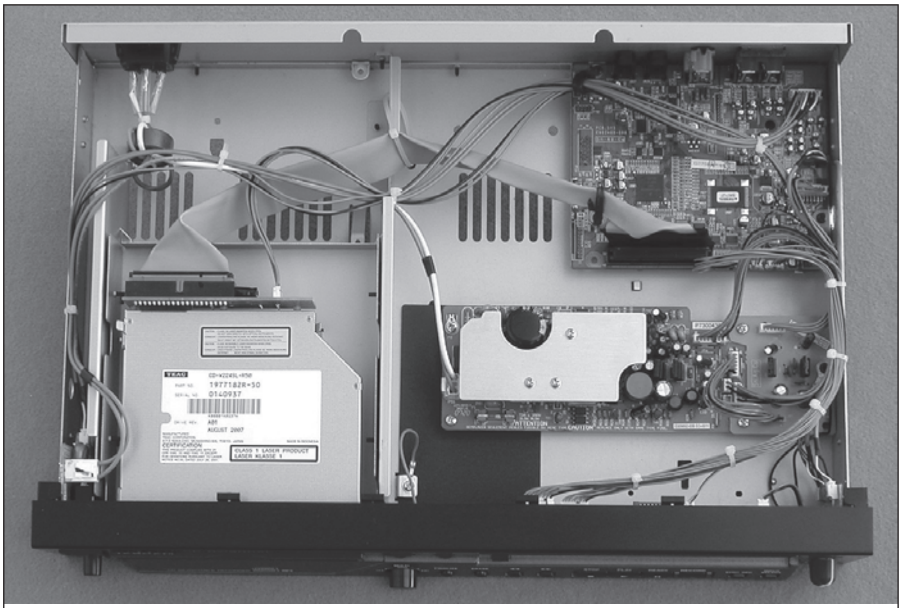

PHOTO 3: Inside view of the CD-RW900SL. Build quality is solid, if not robust. The switching power supply is typical of professional digital recorders.

format, my main digital recorders are a pair of Tascam DV-RA1000HDs, which also allow the connection of a PS/2 keyboard for text entry. The manual for those recorders warns you to turn the power off before connecting the keyboard—otherwise damage to the recorder and/or the keyboard may result. The manual for the CD-RW900SL has no such warning, which seems strange. User beware!

DESIGN DETAILS

For many years the HHB CDR-830 was an industry standard among CD recorders selling for around $600 street price. The CDR-830 was discontinued last fall and the Tascam CD-RW900SL seems to have replaced it as the best-selling CD recorder in this price range. I think it's appropriate to compare the two.

We have seven HHB CDR-830 recorders at The Crane School of Music at SUNY Potsdam, so I have considerable experience with this unit. All ICs in the CDR-830 are mounted on the bottom of the PC board, so I removed the board to investigate. At the heart of the HHB CDR-830 is an AKM Semiconductor AK4524 Audio CODEC, a 96kHz/24-bit combination ADC and DAC chip. AKM Semiconductor, a Japanese company, has been around since 1983, and during the past few years their conversion chips have been used increasingly in affordable pro audio equipment (Behringer uses AKM chips in a number of their digital products).

The AK4524 ADC has single-ended analog inputs and 64× oversampling, with a dynamic range of 100dB, with S/(N+D) specified as 90dB. The DAC section of the AK4524 features 128× oversampling and differential outputs. Dynamic range is 110dB and S/(N+D) is 94dB. AKM also claims high jitter tolerance for this chip.

The three surface-mount dual op amps used in the CDR-830 are all made by NJR: one 4560 for the headphone amp, one 4565 for the analog input amplifier, and one 4558 for the analog output amplifier. Slew rate and gain-bandwidth product are 4V/µS and 10MHz for the 4560; those specs are 4V/µS and 4MHz for the 4565. The rather pathetic 4558 has a slew rate of 1V/µS and a bandwidth of 3MHz, and shouldn't be used in *any* self-respecting piece of audio equipment. Fortunately, the 4558 is only used in playback and won't affect recordings made on the HHB.

The digital core of the Tascam CD-RW900SL is based on three AKM chips (I have a service manual for the Tascam recorder). The AK4528, another 96kHz/24-bit audio CODEC, is used for A/D and D/A conversion. AKM designates the AK4528 as a "high performance" chip, and it appears to be a slightly improved version of the AK4524. All specifications are identical to those of the AK4524 except for dynamic range and S/(N+D) of the ADC section, which are 108dB and 94dB, respectively. This, presumably, translates into improved low-level linearity in the A/D converters. The ADC section also has differential analog inputs.

Input and output sample rate conversion are done with the AD4121A asynchronous sample rate converter. The S/PDIF input receiver and transmitter are also housed in a single combo chip, the AK4113 digital audio interface.

It's strange that the Tascam CD recorder doesn't perform de-emphasis, because all three AKM chips support the Red Book de-emphasis function. All IC op amps in the CD-RW900SL are NJM4580 dual types which, while hardly state-of-the-art, are a distinct improvement over the op amps used in the HHB CDR-830. The 4580 has a slew rate of 5V/µS and a gain-bandwidth product of 15MHz. NJR intended this op amp for audio applications and specifies 1kHz THD as 0.0005% at 5V out into a 2kΩ load (they don't specify THD for any of the op amps used in the HHB recorder). Both the HHB and Tascam recorders have a switching-type power supply, which seems universal in professional CD recorders.

Tascam has taken advantage of the differential inputs of the AK4528. The unbalanced input from the RCA jack is buffered with half a 4580 configured as an inverting amplifier. The next stage, after the input level controls, consists of two halves of a 4580—both operated in the inverting mode—converting the single-ended signal to true differential. This differential signal is fed to the analog inputs of the AK4528's differential inputs.

Strangely, Tascam seems to have "tacked on" the balanced inputs on the CD-RW901SL version of this recorder. The balanced inputs are converted to single-ended with a differential amplifier—half of a 4580 per channel—and then fed to the single-ended input level controls and converted back to differential using the same circuitry as the CD-RW900SL. Performance would be improved, and the signal path simplified, if Tascam had made the signal path fully differential from the XLR jacks to the analog inputs of the AK4528.

So, what do you get in a state-of-the-art digital recorder, in comparison to CD-RW900SL? I have been using a pair of Tascam DV-RA1000HD digital recorders for live concert recording at The Crane School for about two years—they replaced our third-generation Sony PCM-R500 DAT recorders. The DV-RA1000HD uses state-of-the-art Burr-Brown/TI converter chips—a PCM1804 A/D and a DSD1792 D/A converter. Differential current-to-voltage conversion is accomplished with a pair of OPA2134 op amps, a device well-known in high-end audio. All other op amps are New Japan Radio NJM2114M, a high-performance chip several notches above those mentioned (or the still-popular 5532), with a slew rate of 15V/µS, gain-bandwidth product of 15MHz, and high output current of 60mA.

The DV-RA1000HD records onto a 60GB hard drive at 24-bit resolution and all standard sampling rates from 44.1 up to 192kHz. The hard drive is intended only as a temporary holding area. You can archive 44.1kHz recordings to the optical drive as a standard Red Book CD, or copy individual tracks as files to a DVD. Recordings made at higher sampling rates are archived to DVD.

The DV-RA1000HD can also record in Direct Stream Digital format. DSD recordings can be archived to DVD, but the recorder won't make SACDs, and the exorbitant cost of DSD editing and SACD mastering software make the DSD capability a moot point for me. Even the high-end DV-RA1000HD has not escaped the use of switching power supplies.

SONIC PERFORMANCE

I've found the Tascam DV-RA1000HD to be a reference digital recorder. It is the most natural-sounding digital recorder I've used—extremely detailed, with a smooth treble region, extremely low noise floor and a wide, spacious soundstage. These qualities are apparent even at 44.1kHz, and get even better at higher sampling rates.

Red Book CDs archived to its internal optical drive are sonically superior to those made on *any* CD recorder I've used, and it's also miles ahead of the various DAT recorders I've used over the past 18 years. The CD-RW900SL doesn't match the performance of the DV-RA1000HD, and it shouldn't be expected to. But, it certainly offers a noticeable sonic improvement over the HHB CDR-830.

For comparison to the HHB CDR-830, I recorded excerpts from three of my favorite high-resolution digital discs on the HHB and the Tascam recorders (**Table 1**). These carefully prepared discs are among the finest in my collection sonically, and are probably as close to rolling the original analog tapes in my listening room as I'm likely to get. (I believe that the Rachmaninoff *Symphonic Dances* DVD was actually made from a 15-ips copy that engineer David Hancock prepared from his original 30-ips tape, but it's still a knockout.) I played these high-res discs back on my reference multi-format player, a recently-purchased NAD M55, and fed the tape outputs of my reference preamp (my redesigned Adcom GFP-565) to the analog inputs of the CD recorders.

After making the test recordings, I extracted them to my digital editor (Sony's Sound Forge 8.0), carefully normalized levels on each track individually to ensure level matching between the HHB and Tascam recordings, and burned the results to a new CD-R (using Sony's CD Architect 5.2). I then compared the HHB and Tascam tracks by playing this CD-R on the NAD M55.

Both recorders shrink the soundstage somewhat, but the Tascam fares better than the HHB. Localization within the soundstage is more precise on the Tascam, and comparatively vague on the HHB. Some of the HHB's soundstage shortcomings may be related to its somewhat dry sonic character. The Tascam captures hall ambience better, which results in superior reproduction of the spatial qualities of the test recordings.

The HHB adds a layer of grain to the treble, particularly on orchestral violins and has a somewhat "tizzy" quality on cymbal crashes. The Tascam is not the ultimate in high-frequency smoothness, but it is noticeably better in the treble than the HHB. The Tascam also captures inner detail better than the HHB, and orchestral climaxes become less congested on the Tascam. In the bass end of the spectrum, the Tascam is slightly fuller and weightier than the HHB.

To get the most out of recordings made on the CD-RW900SL, I recommend using an outboard D/A converter, which most professional users will probably have. As a stand-alone player, the Tascam is decidedly mid-fi. I found it rather thin and somewhat dry sounding, with a rather narrow soundstage, and not particularly impressive in revealing inner detail. The liquid, transparent, three-dimensional sound that I'm used to with the Monarchy M24 DAC (reviewed in *aX* Oct. 2007, with measurements by Chuck Hansen in the June 2007 issue) just isn't there with the CD-RW900SL as a stand-alone. It does make a very respectable transport, however, when used with a high-performance outboard DAC. The CD-RW900SL is a better-sounding recorder than stand-alone player.

CONCLUSION

The Tascam CD-RW900SL is a solid performer in its price range. Though its performance falls short of the state-of-the-art in stand-alone digital recorders, it makes audibly superior recordings to the previous generation of CD recorders, specifically the HHB CDR-830. The pitch

TABLE 1: High-Resolution Reference Discs

Rachmaninoff: *Symphonic Dances*, Op. 45. Dallas Symphony Orchestra conducted by Donald Johanos. Classic Records DAD 1004. (96kHz, 24-bit PCM transfer of 30-ips analog tape, or possibly a 15-ips copy, engineered by David Hancock.)

Rachmaninoff: Piano Concerto No. 3. Byron Janis, pianist; London Symphony Orchestra conducted by Antal Dorati. Mercury Living Presence SACD 470 639-2.

Stravinsky: *The Firebird*. London Symphony Orchestra conducted by Antal Dorati. Mercury Living Presence SACD 470 643-2.

control should be redesigned to eliminate the high-frequency rolloff and increase in distortion. I also prefer a conventional tray-loading mechanism, and agree with Chuck Hansen's concern about the possibility of scratching CDs.

Although I had no problems in this regard on the recorder I reviewed, another CD-RW900SL that we purchased at The Crane School occasionally makes hairline marks on discs (contact Tascam if you have this problem). But, all things considered, those in the market for a CD recorder in this price range will find the CD-RW900SL a good choice. ■

Tascam CD-RW900SL—Measurements

By Chuck Hansen

First, a couple of operational squawks about the Tascam CD-RW900SL CD Recorder. I don't particularly care for the slot-loading CD drive that Tascam used. The CD gets dragged through two plastic lips when it loads. If there is any dirt on them, they can scratch the CD. And if a CD becomes trapped inside, there is no pinhole to insert a paper clip and release the tray. The troubleshooting section of the 36-page manual tells you what to do if you can't insert a disc, but there is no remedy listed if a disc won't eject. I prefer open-tray-style computer drives.

Second, Gary Galo and I needed to troubleshoot the initial poor high-frequency response and high distortion problem I found during my initial tests. We finally noted that the variable pitch function had been left on, although set to 0.0%. My THD test set had no problem syncing to the test CD frequencies because pitch was not set off-frequency. The fact that it messes up both frequency response and THD makes it unacceptable to me, negating any benefit it might otherwise have. That "feature" ought to be restored to OFF whenever the unit is turned off, so you get Pitch Off when you power up again.

MEASUREMENTS

I operated the CD-RW900SL for one hour with the Pierre Verany test CD before making any measurements. I performed tests using the CBS Labs CD-1 and Pierre Verany test CDs (both available from Old Colony Sound Lab). I recorded tones onto 16-bit 44.1kHz CD-Rs from the oscillator in my distortion test set to check the recording function.

In order to check the MP3 playback, I used a custom CD-R I made by ripping analog test tones to 128k sampled MP3 files. I evaluated the CD-RW900SL digital output by connecting it to the 24/96 DAC input in my Alesis MasterLink ML-9600. **Table 1** summarizes the measured performance and the manufacturer's specifications.

The capacitor-coupled unbalanced RCA analog input impedance measured 24k at 20Hz, and 22.5k at 1kHz and 20kHz. The capacitor-coupled unbalanced RCA analog output impedance at

TABLE1: MEASURED PERFORMANCE

Parameter	Manufacturer's Rating	Measured Results—Variable pitch off (see text)
bit depth/sampling frequency	16-bit, 44.1kHz	verified
frequency response (44.1kHz) 20Hz-20kHz:	±0.8dB (playback) ±1.0dB (recording)	+0.23/-0.01dB (playback) +0/-0.56dB (recording)
signal-to-noise ratio and dynamic range:	95dB (playback) 90dB (recording)	96dB (playback) ref 2V RMS, A wtd
Total harmonic distortion:	0.006% (playback) 0.008% (recording)	0.0030% (playback) 0.0037% (recording) 1kHz, 0 dBFS
IMD – CCIF (11kHz + 12kHz and 19kHz + 20kHz):	N/S	-68dB (0.039%), 1kHz -65dB (0.056%), 1kHz
channel separation:	90dB (Playback) 80dB (recording)	96dB (playback) ref 2V RMS, A wtd
wow and flutter:	<0.001% (unmeasurable)	N/A
Analog input impedance:	22k ±10%	22.5k 1kHz
Analog output impedance:	600Ω ±60Ω	586Ω 1kHz
Phones output:	20mW 32Ω	24mW 32Ω 1kHz
Digital I/O:	coax RCA 75Ω, optical Toslink	76.1Ω (input)
Power requirements:	17W	N/A

20Hz was 836Ω, decreasing to 586Ω at 1kHz and 20kHz. Balance between the two channels was essentially perfect. The headphone jack output impedance measured 31Ω at 20Hz and 1kHz, and 39Ω at 20kHz. It delivered 24mW maximum at 1kHz. All analog outputs had normal polarity, a positive-going pulse producing a positive-going analog output.

The S/PDIF digital input presented the appropriate 75Ω impedance, while the digital output jack had a very high resistance. This may indicate that the digital output is capacitively coupled.

The DAC frequency response showed a slight rise of +0.23dB from 13kHz to the point where the digital output filters sharply rolled off the response (**Fig. 1**). The 0dBFS indicated unbalanced analog output was 2.04V RMS at 1kHz, or about 0.18dB above the CD Red Book standard of 2V RMS. The dB level display changed in 0.05dB increments in response to the analog input level controls.

I also measured the frequency response when the variable pitch control was on and set on-frequency (no pitch change commanded). This is the curved line at the HF end of **Fig. 1**, measured with a -0.05dBFS analog input signal (to avoid digital overload). The dash-dot line is the MP3 frequency response.

Crosstalk between channels and the playback signal-noise ratio at 1kHz were both 96dB referred to the Red Book 2V RMS. The headphone output at its maximum volume control setting (32Ω load) was 24mW at 20Hz and 1kHz, decreasing to 18.4mW at 20kHz. The maximum open-circuit headphone output voltage was 2.93V RMS at 1kHz.

The record-reproduce frequency response compared to the -0.05dB flat analog input signal is shown in **Fig. 2**. The top curve at 15kHz is the CD-R playback of the recorded signals. The dashed curve is the response to the analog input signal as measured at the analog output jack with the CD-RW900SL in Record/Pause mode. The lower curve at 15kHz

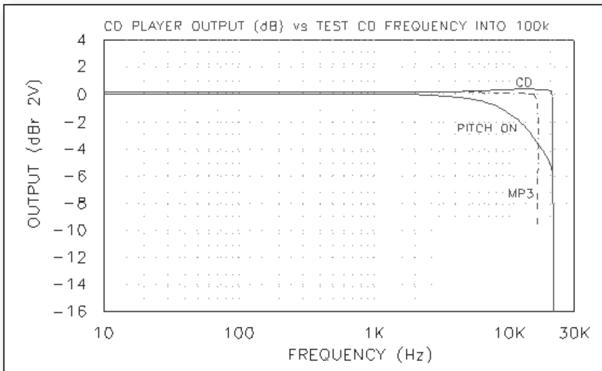

FIGURE 1: Frequency response.

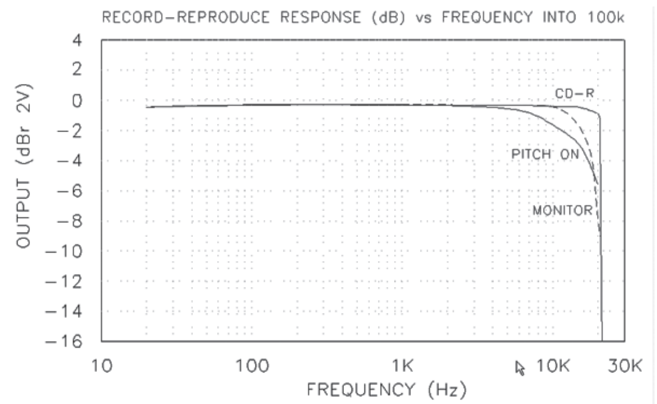

FIGURE 2: Record/playback frequency response.

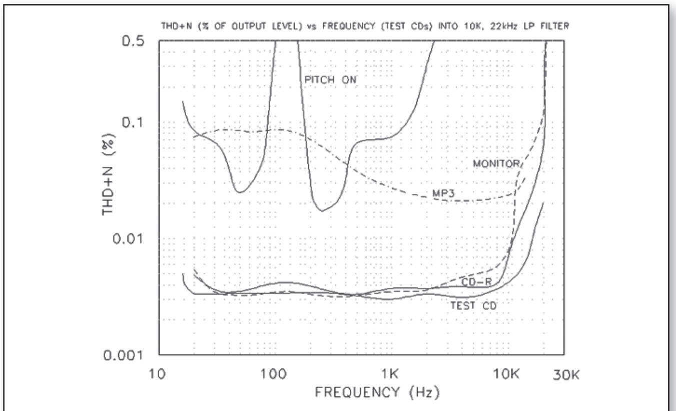

FIGURE 4: THD+N vs. frequency.

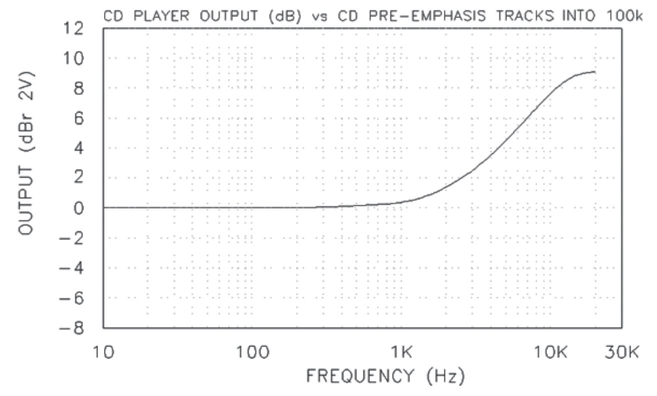

FIGURE 3: Pre-emphasis error.

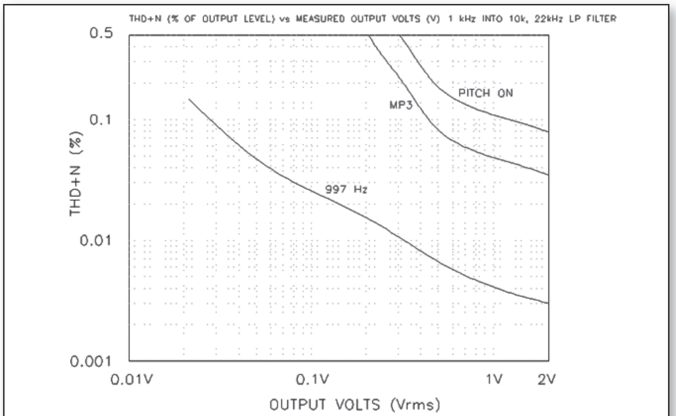

FIGURE 5: THD+N vs. output voltage.

is the CD-R playback response with the variable pitch control on with no pitch change commanded.

Playing the CD-R on my Alesis ML-9600 verified that the CD-RW900SL had accurately recorded the analog signal. Similarly, the CD-RW900SL made an accurate digital recording of selected Test CD tracks played from the ML-9600 S/PDIF digital output. It also showed flat response with its digital output connected to the 24-bit 96kHz DAC in the ML-9600 while I played the Red Book Test CD on the CD-RW900SL. There is no MP3 recording capability with the CD-RW900SL.

Figure 3 shows the pre-emphasis error when I played pre-emphasized digital test tracks. The CD-RW900SL does not appear to recognize the pre-emphasis flag on the test tracks.

THD+N versus frequency is shown in **Fig. 4**. I engaged my distortion test set 22kHz 4-pole low-pass filter to remove out-of-band noise. The Red Book Test CDs produced the lowest level of distortion, followed by the CD-R of the analog test signals. The dashed line is the record

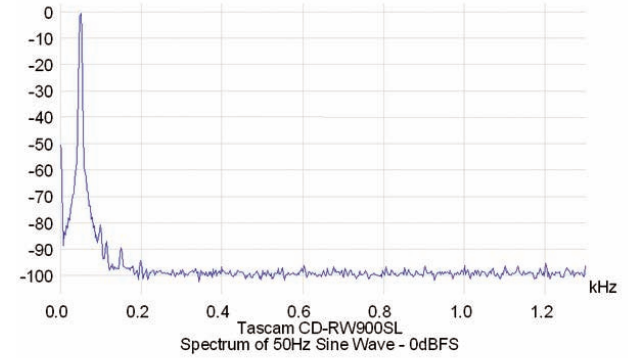

FIGURE 6: Spectrum 50Hz sine wave.

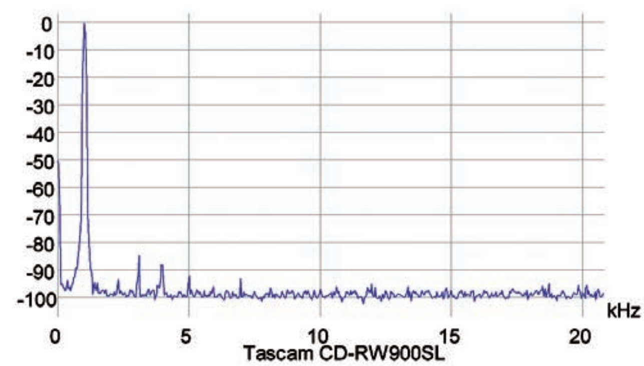

FIGURE 8: Spectrum 1kHz sine wave.

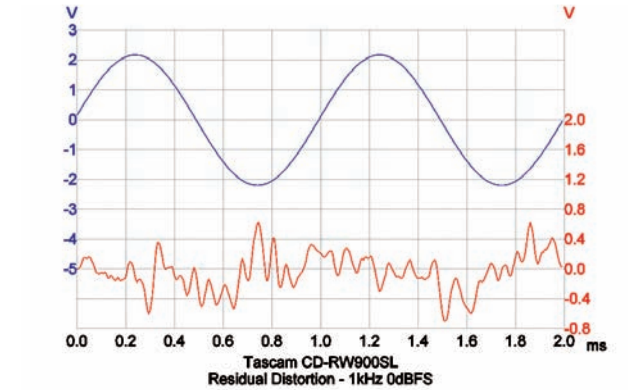

FIGURE 7: 1kHz THD residual.

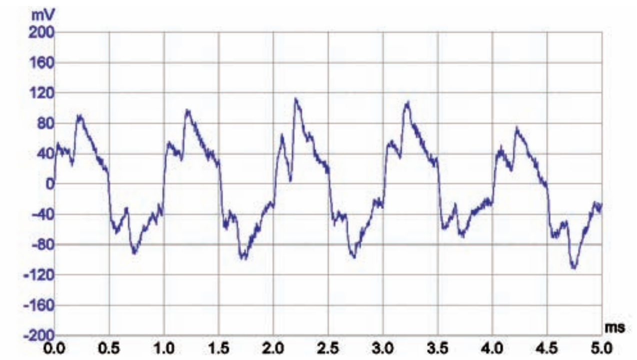

FIGURE 9: -90.31 dBfs 1kHz sine wave.

monitor signal at the analog output jack with the CD-RW900SL in Record/Pause mode. The dash-dot curve is the THD+N produced during the MP3 test track playback.

The wildly-varying upper curves represent the large increase in distortion with the variable pitch control on and with no pitch change commanded. The THD reached nearly 3% at 127Hz and is off the upper scale on the graph at that point. I also measured 1.3% at 4kHz, 5.4% at 8kHz, and 7.9% at 10kHz. It continued to increase in this fashion until it exceeded 18% at 20kHz.

The lower curve in **Fig. 5** shows the THD+N versus output voltage using the 997Hz decreasing amplitude signal on the CBS test disk. There is no clipping evident because of the CD test signal source. You can, however, drive the CD-RW900SL into clipping from the analog inputs. When I overdrove the unbalanced analog input, the distortion went straight up just above the point where the 0dB level indication occurred.

The middle curve is the THD+N ver-

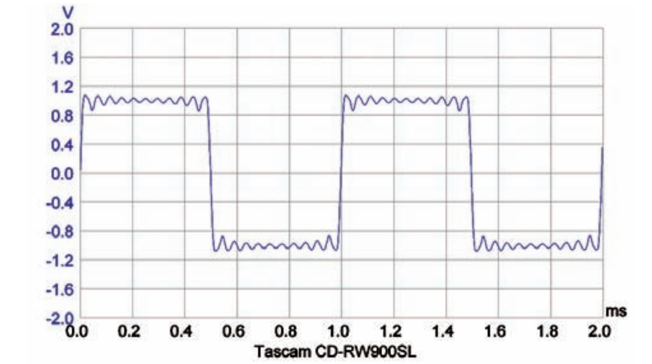

FIGURE 10: 1kHz square wave.

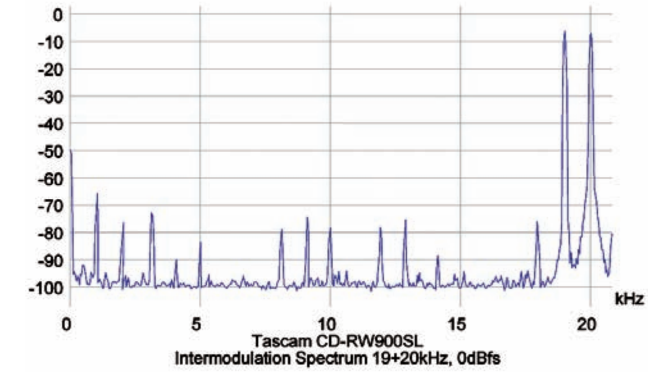

FIGURE 12: 19kHz + 20kHz IMD.

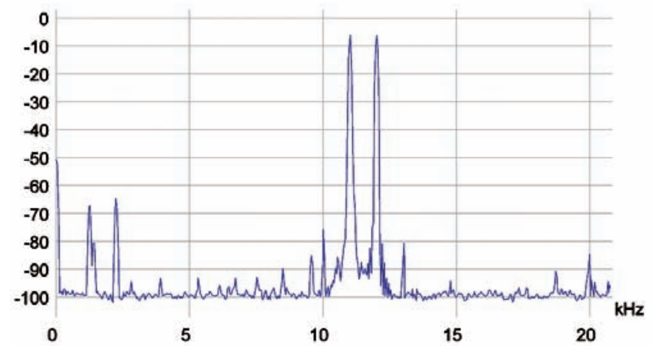

FIGURE 11: 11kHz + 12kHz IMD.

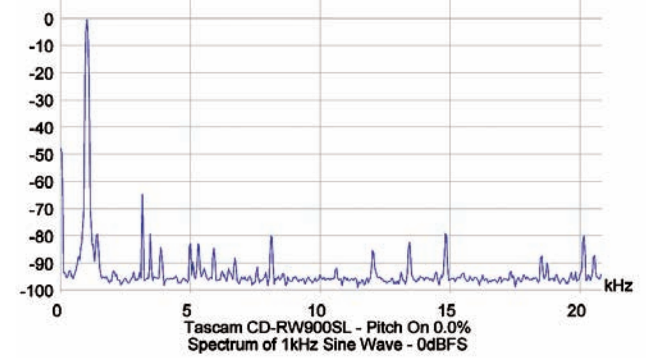

FIGURE 13: Spectrum 1kHz sine wave—pitch on 0.0%.

sus output for MP3 playback files, and the upper curve represents the test CD tracks when the variable pitch control is on with no pitch change commanded.

The spectrum of a 50Hz sine wave at 0dBFS is shown in **Fig. 6**, from DC to 1.3kHz. The calculated THD+N based on the first five harmonics is 0.0022%, with no significant harmonics and only low-level noise artifacts. The measured THD+N is just 0.0034%.

The distortion residual waveform for a 1kHz sine wave at 0dBFS is shown in **Fig. 7**. The upper waveform is the analog output signal, and the lower waveform is the monitor output (after the THD test set notch filter), not to scale. This distortion residual signal shows mainly lower order harmonics. THD+N at this test point is 0.0027%.

I also performed a spectrum analysis at 1kHz 0dBFS, as shown in **Fig. 8**, where THD+N measured 0.0030%. The most notable harmonics are 3^{rd}, 4^{th}, 5^{th}, and 7^{th}.

Figure 9 shows the reproduction of an undithered 16-bit/44.1kHz 1kHz sine wave at –90.31 dBFS. At this level the signal consists of ±1 bit of data, produc-

Continued on p. 39

Continued from p. 37
ing two different voltage levels that are symmetrical about the horizontal (time) axis. These discrete voltage steps are indistinct and have a triangle shape rather than the expected flat levels.

Figure 10 shows the response of the CD-RW900SL to a 1kHz 16-bit 44.1kHz square wave from the CBS test disk. It shows the characteristic Gibbs Phenomenon ringing associated with the limited bandwidth produced by the digital filters. The ringing is nicely symmetrical with no sign of analog clipping at the peaks.

Figure 11 shows the spectrum response to equal level of 11kHz and 12kHz signals, each at –6dBFS, from DC to 20.8kHz. There are significant intermodulation products at 1kHz, 2kHz, 10kHz, and 13kHz, with the 1kHz product measuring -68dB (0.039%). This indicates a degree of nonlinearity somewhere in the reproduction chain of the CD-RW900SL.

Figure 12 shows the spectrum response to equal level of 19kHz and 20kHz signals, each at –6dBFS, from DC to 20.8kHz. The intermodulation products appear almost like a picket fence, showing higher nonlinearity as the IMD frequencies increase. The 1kHz product is -65dB (0.056%).

The CD-RW900SL ignored defects on the Pierre Verany Test CD-2 out to track 33. At track 34 (2mm defect) the unit cut off for a second after passing through the defect, going totally mute at track 36, which represents a large 2.5mm defect. The unit easily met the Red Book requirement of 0.2mm maximum.

Finally, **Fig. 13** shows the spectrum of the 1kHz sine wave at 0dBFS with the Pitch control On and set for 0.0% pitch change. Distortion measures 0.074%.

The Tascam CD-RW900SL is a serviceable CD recorder. The digital recording capability is quite good, with its main limitation in what I assume is the analog sections, as shown by the pronounced increase in THD at the higher frequencies, and the nonlinearity when reproducing the two sets of CCIF intermodulation tests.

The Variable Pitch function produces high levels of distortion and a premature rolloff in the high frequencies. Use it at your own sonic peril! *aX*

Yard Sale

Wanted

One Adcom power amplifier GFA 535, reasonable condition.
L. Campbell 1-941-697-2669

Tubes: 6EU7, 6BM8, 6AN7 (X3), 6BY8, 6AS8, 6AM8A, 6AD8, 6DC8, 7125, 6AQ5.
All must test good or new. For project.
Craig Kendrick Sellen
Highland Manor
164 South Main St.
Carbondale, PA 18407-2655

NOS Stanton 680 Hi-fi replacement stylus without brush in plastic cube w/ litt..; Bituminized felt (about ½″) for loudspeaker project. NOS RCA original mono LM 1779 Victory at Sea Red Seal label. Good CD or cassette copy OK.
Vincent Mogavero 718-672-7478. vmogavero@aol.com.

Hi-fi and stereo equipment (tubes and transistor, kits and factory built): Heathkit, Eico, Scott, Dynaco, and so on. Turntables, records, reel-to-reels, tapes. I am a collector, not for resale.
Please e-mail me at njvet01@yahoo.com.

For Sale

Pair ScanSpeak 18W/8546-00 6.5″ Kevlar MidWoofer $150
Pair ScanSpeak 18W/8545-00 6.5″ carbon fiber MidWoofer $175
Pair Vifa D27SG05-06 shielded 1″ fabric dome Tweeter $25
Pair Vifa TC26SF05-06 shielded 1″ fabric dome Tweeter $20
edgiaimo@comcast.net

TEAC A-2300SX reel-to-reel 7″ 7.5/3.75IPS $150
Technics SL-1800 Turntable 33/45RPM no cartridge $125
AUDIO Magazines Jan. 1978 through Sept. 1996
Buyer arranges and pays shipping for above items or pickup Minneapolis area.
Bill at 763-786-2692 or bwerner@mnnradio.com

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Ad Index

PHONO FIX

I want to thank Gary Galo for his article, "A Belt-Drive Turntable Upgrade," (July '08 *aX*, p. 20). I found his instructions extremely helpful. It just so happened that I was fixing up an old Thorens TD125 MKII turntable with a Rabco linear tracking tonearm (**Photo 1**). This LP rig had been sitting idle for some time and was in need of repair.

Like Gary's turntable, the TD125 MKII is also a belt-drive turntable with a suspended sub-base. And, hooking up a cable of any size to the phono outputs on the Rabco tonearm caused the tonearm to hit the edge of the turntable base.

So, I took Gary's advice and attached a small metal plate to the base with new RCA jacks (**Photo 2**). I also routed some Cardas shielded tonearm wire and a separate ground wire through two small holes I drilled in the bottom of the turntable. The Dayton Isolation Cones he suggested provide a solid foundation and worked wonderfully for leveling my turntable.

An additional problem I ran into was bearing noise. I heard a grinding sound coming from the bearing and platter spindle when I spun the platter. I removed the old bearing oil, cleaned the bearing, and added some 30 weight synthetic motor oil. This didn't get rid of the noise, however.

I decided it needed a higher viscosity oil. So, I used Mobil One 75W - 140 weight synthetic gear oil. For extra insurance, I also added 25% of Prolong oil treatment to the oil. That fixed it good! I heard no audible bearing noise and it spun forever.

I found a perfect replacement for my worn turntable belt at Microphonics (http://stores.ebay.co.uk/microphonics).

PHOTO 1: Thorens turntable.

PHOTO 2: Phono fix.

The turntable sounds great. I'm now enjoying my LPs again and buying more. Thanks.

John Loudenback
johnloudb@earthlink.net

SPEAKER ADVICE

As I scanned the price list in the Loudspeaker Buyer's Guide (*aX* August '08), I was taken aback by the high prices. However, the high prices go hand-in-hand with the high quality the speakers offer. These are not cheap speakers that you'll find in chain electronic stores. If you give them a listen, it will become clear that a lot of thought and care went into their design and construction. You must also remember that it costs big bucks to build just about anything in the US these days, due to high labor costs and other issues.

While you can understand the high costs of the new speakers, in these difficult times, they also put the dream of owning a great pair of speakers on hold for many of us. Until you win a hefty jackpot in your nearest casino, inherit money from a rich relative, or maybe dig up the loot that got away from D.B. Cooper in 1971, you have other options.

Many folks, including myself, are buying the Radio Shack 40-215 15″ PA speaker. I paid $100 each, but now they sell for $80, at least in the San Francisco

Bay Area. Though many might shy away from their black-carpeted cabinet, these guys really pack a wallop. I find the mids and highs very impressive as well. Classical music and jazz listeners might find problems, but rockers and rappers will love 'em. Those interested would do well to act quickly, because some stores have taken them off display—which could mean they could soon, if not already, be discontinued.

If you miss out on the Radio Shacks, you can check out your local used hi-fi retailer(s), which you can find in the Yellow Pages. The 2008 *World Tube Directory* (available from *aX*) lists a number of such stores.

Thrift stores occasionally have nice speakers, though you might need to make repeated trips. The better known chain thrift stores receive their merchandise from an area distributor. Because of this, you will find very few electronic items, if any. Independent thrift stores, however, might have a lot of used audio gear—and many speakers. A good pair might cost from $20 to $100. Before you buy them, remove the grille and check for punctures and rotted or missing foam surround.

I will simply replace damaged or blown speakers or grilles, and repair damaged cabinets. If you replace a speaker in one enclosure, you would do well to also replace the same speaker in the other enclosure, for consistency in frequency response, even if the other speaker is good. Put the good speaker in a plastic bag (to protect it from mice and insects) and store it in a cardboard box for future use. Past issues of *aX* have some great articles on speaker upgrades and repairs that might prove helpful. Old issues of *Speaker Builder* will also prove helpful as well.

Good buys on speakers show up in your local newspaper's classified ads, but be careful, because many stolen items appear. If you think you're paying too little, then you might be buying "hot" speakers. Never buy speakers with missing serial numbers, especially from a stranger. It's better to buy from a store, which is what I do. Happy hunting!

Neal A. Haight
Castro Valley, Calif.

HEAR, THERE AND EVERYWHERE

I received an e-mail from Ed Dell right after I'd written to him about the English DIY company debacle. It seems that a British "watchdog" group lambasted them for saying that cables could sound different from each other. The funny part of this is that the group didn't lambaste the big cable companies, they picked on a small DIY cable company that advertised how to copy (fill in the blank brand) expensive cables. It seemed odd that the group wouldn't attack the real cable companies, only a small DIY company, and that an *audioXpress* reader, who should have known better, being a DIYer himself, was agreeing with the watchdog group (see July and Nov. isues).

So, Ed Dell asked if he could print my letter, and not really wanting to get involved in this heated debate, I basically said no. But after receiving the latest issue of *audioXpress*, I found another nasty letter to the editor from another DIYer who can't hear any difference in anything. That's like singling out the Triumph Motorcycle Company for not putting "you should wear a helmet" in their advertising, and not even mentioning that the giant motorcycle manufacturers (Kawasaki, Suzuki, Yamaha, and Honda—the big four) also failed to mention wearing a helmet in their ads, but were not chastised. Then folks starting to write nasty letters to the "motorcycle modifier" magazines, complaining about the terrible sin that Triumph committed.

To make a long story even longer, here is what I have noticed about the sound of different audio components. First, let's take money totally out of the equation. That seems to be what makes people the angriest. Have you ever heard the old saying, "God must not care about money, after all, look at the people he gives it to."

I have an ancient Heathkit preamplifier. I've had it so long and I've stripped it down so much that I can't even recall the model number. It was burnt orange with chrome accessories, much like a 1950s car dashboard. I have never modernized the design of this in any way.

I've stripped down the preamp by removing the tone control circuits and tubes; I've removed the big rat's nest of selector and installed a two-position selector from my junk drawer (i.e., phono/aux). I've gotten rid of the stereo/mono reverse rat's nest. So, right now, it's a tube rectified tube chassis with a line stage and quite good MM phono stage and the original volume control. The tubes are Mullards, from the 50s. They've never gone bad—I've never needed to change them.

I also have a late model, solid-state preamplifier that cost quite a bit, despite the fact that I got a good deal on it. If you remove the top, it's laced with expensive capacitors, diodes, resistors, and so on, and a very expensive volume control. Without mentioning names or prices, this preamp has received warm reviews from most all the audio magazines.

In my listening room, I have both preamps set up so that I can easily switch between the two by changing the patchcord. Everyone who has ever entered my listening room has taken the following listening test. At the exact same volume, set by a sound pressure gauge, I switch between the preamps, playing various kinds of music. The results are simple, non-mathematical, and resounding: every single person who has ever taken this test has picked the Heath preamp. I am talking about at least 30 people over the years. They all said basically the same thing. The audiophiles used phrases such as "soundstage was better," "warmth," and "bloom." The non-audiophiles also picked the Heathkit—they said it just sounded "way better."

When my family and I listen to solid-state, after an hour or two, we usually stop to do something else. But when I hook up the stereo with the tube-regulated power supply Heath preamp into two tube-regulated power amps, we'll start listening to music on Saturday morning during breakfast and will be hard-pressed Saturday night at 2am to turn off the stereo and go to bed. I don't think you can measure that! If you can measure it, and there's something wrong with the tube measurements, I'll take it anyway. Like the song says, "if loving you is wrong, I don't want to be right."

Clayton Mitchum

BUFFER USE

I have a question for Darcy Staggs about his article, "Inductive RIAA Compensation," Feb. 2003, p. 40. Why do you need a buffer after the OPA627? My experience is that the 627 can drive a fairly difficult load. If you do need the buffer what would you recommend?

Kit Fung
fung1580@rogers.com

Darcy Staggs responds:

Thank you for your interest in inductive RIAA compensation[1]. My version is taken from the preamp article by Michael Danbury, who also assisted in its adaptation to my phono stages.

At the time I performed the changeover from capacitors to inductors, I was using a highly modified Hafler DH-110. It had an added op amp phono stage using the Texas Instruments TL-070, which sounded better with a simple buffer (designed by Mike). I used the same buffers on the dual op amps in my Magnavox CD-472, which improved the sound. The phono stage buffers remained in place when the OPA-627 was swapped in, so I showed a buffer in the article, partly for generality.

Now I am listening to Preamp X[2] with another op amp in the phono stage, still using the inductive RIAA solution, but without any buffer. Newer op amps have been tested in Preamp X and all seem to drive the volume and balance control, and so on, with no buffer needed, but then again every preamp will be different and some may in fact benefit from a buffered phono stage.

1. *Audio Amateur*, 3/96, p. 8.
2. *audioXpress*, 4/07, p. 46.

HOW CDS WORK

I am writing this prompted partly by Edward T Dell Jr.'s "review" of Mystery Cream in the Sept. 2008 issue, and partly by my own annoyance at the misinformation that persists in this area—and the way it never seems to get corrected.

Audio CDs contain numbers (44,100—16 bit numbers per channel per second). Depending on which type of CD, these numbers are recorded in a series of dots burned into the surface of a layer of dye using a laser (CD-R), or as bumps or pits—depending on which way you look at it—on the surface of a layer of metallized plastic (pressed commercial CDs). In both cases, they are read by a laser, which is bounced off the data surface and, with the aid of some neat optics, reads the pits or dots as bright and dark spots in the reflected beam. The surface which the laser is reading is inside the disk, and is covered by a layer of clear plastic to protect it—the laser is focused through that clear cover onto the data layer.

(I'm skipping a lot of stuff about how the beam follows the tracks...)

The way virtually all players work is that the data is held in a buffer, which is a queue where the numbers are stored. The amount of data in this buffer is controlled by a servo mechanism such that the buffer is never allowed to run empty. When the amount of data in the buffer becomes too low, the mechanism reads more data off the disk. Numbers are read *out* of the buffer and fed to the digital-to-analog conversion circuitry (D-to-A converter) under the control of a crystal controlled clock.

Unlike a turntable, the speed at which the disk spins is *completely unimportant* as long as the buffer isn't allowed to run empty. In fact, most mechanisms read the data in short bursts, pausing in between, and do so at different speeds on different portions of the disk. It doesn't matter whether the disk vibrates, bounces, or even stops, as long as the player mechanism can manage to follow the track and deliver enough numbers to keep the buffer supplied. Many portable players are able to find the proper track again, even after being bounced out of it entirely, before the buffer runs empty—and continue to play uninterrupted.

The numbers coming out of the buffer, under control of the clock, are fed to the D-to-A converter. The *only* thing that determines the jitter associated with those numbers at that point is that clock. The speed of the disk's spin, or how smooth it is, means *nothing*. Neither do things such as the disk vibrating or edge reflections, or resonances, and so on. (In theory it would be possible that a clock was somehow influenced by what the rest of the circuitry was doing—in practice the jitter introduced by this clock is always determined simply by how well the clock itself is designed.)

Jitter is important because at this point it will cause most D-to-A converters to produce distortion (many of the better separate ones have their own ways of preventing this problem). All of this assumes that the disk mechanism is indeed able to read correct numbers—but what if the numbers aren't all correct?

Audio CDs contain basically three error correction mechanisms. When the numbers are recorded, extra information is added in case they become damaged. This extra information, and the data itself, is "spread" in such a way as to minimize the chance that surface damage will cause enough data to be lost that it cannot be repaired. Before the numbers leave the final buffer to head off to the D-to-A converter, they are processed digitally to verify that they are correct and to repair any that are not.

The first two error correction mechanisms are absolute. They can and will detect and repair *any* error perfectly, ranging from a single digit to a gap 2.5mm wide in the disk. (This is assuming that the drive mechanism can follow the track—some fail to maintain tracking before this point is reached.) Any error repaired at this stage *no longer exists*—you can't possibly hear it because the numbers have been returned to what they should be.

The final correction mechanism uses "interpolation," which is a last ditch attempt to correct any gaps that are too big to be properly corrected by the first two levels, and is accomplished by filling them in with a "good guess" of what belongs there. Errors of this level often make "ticks" and most certainly *will* affect the sound of the disk—especially if there are a lot of them—because the original numbers have been lost and cannot be replaced correctly.

To summarize so far, anything that happens up to this point is totally *meaningless* as long as the buffer remains full, and as long as no errors occur that are beyond the ability of the two digital error correction levels to repair. Unless something *fails*, the numbers coming from a $15 Walkman and those coming from a $25k super titanium gonzo CD deck playing a good disk will be *the same* at this point...and *being the same numbers they will sound the same*.

(I will add a qualification here: I have heard claims that some cheap Walkman-type players actually fail to buffer the full 16-bit numbers that compose the audio—and so do *not* deliver the correct numbers. I've never seen one like that but, if true, this is limited to a few *very* low-end portable players—which probably don't have digital outputs anyway.)

The quality of the clock used to send the numbers out to the D-to-A converter may matter, and the cabling, if the two units are separate, may introduce jitter at this point, which could also matter. Certain high-end D-to-A converters are immune to this jitter as well, but some are not. Separate boxes that remove jitter at this point also exist.

So, it sounds as though I'm saying that "magical mystery cream" and its brethren couldn't possibly work, right? Not exactly.

Those two levels of digital error correction can repair a *single damage* up to 2.5mm across, but most beat-up CDs have a bunch of small scratches and not one big one. Due to the way the data is spread, enough small scratches—especially if they happen to fall in the wrong spots relative to each other—can cause damage sufficient to make the digital error correction unable to totally repair it. In that case, we are stuck with level 3 repair (interpolation), which can sometimes be heard—and can sound quite bad. Theoretically, a very badly vibrating disk, or a very badly designed mechanism, could also cause the drive to return so many bad numbers that they are beyond perfect digital correction.

If that miracle cream is able to fill in some of those scratches enough that the laser can again see through them clearly enough to read more numbers correctly, or if that disk clamp can prevent the disk from vibrating, and so allow that poorly designed mechanism to read a few more numbers accurately, it could make the difference between a perfect play and a less-than-perfect play—but only *if* those conditions exist.

Therefore, any claim that a product will somehow "enhance" the way an undamaged disk sounds when played on a good player is a complete impossibility. What is possible is that some product may enable a borderline disk to play perfectly, or enable a borderline player to play a few more disks perfectly. It may

CONTRIBUTORS

Bill Fitzmaurice ("The XF210 Electric Guitar Cab," p. 8) has been a professional musician since 1966 and has been constructing instruments, amplifiers, and speakers for just as long. Vice president of DeltaSounds Loudspeakers Inc., Fitzmaurice is the author of Speaker Builder's *Loudspeakers for Musicians* and over 30 magazine articles dealing with speakers and electric instruments, as well as the action-adventure novel *Operation: Sergeant York.* Bill and his wife of over 30 years reside in Laconia, NH.

Charles King ("A Really Simple PAS Modification," p. 17) resides in Oneonta, NY.

Earl Geddes ("How Horns Work Revisited," p. 18) runs GedLee LLC with his wife Lidia Lee. He has authored several books, and was recently interviewed in *Voice Coil* (Oct. '08).

Ed Simon ("2400W of Clean AC Power, Pt. 4," p. 24) received his B.S.E.E. at Carnegie-Mellon University. He has installed over 500 sound systems at venues including Jacob's Field, Cleveland, Ohio; MCI Center, Washington D.C.; Museum of Modern Art Restaurants, New York; The Coliseum, Nashville, Tenn.; The Forum, Los Angeles; Fisher Cats Stadium, Manchester, N.H.

Gary Galo (Product Review: Tascam CD-RW900SL CD Recorder, p. 28) is Audio Engineer at The Crane School of Music, SUNY Potsdam, where he also teaches courses in music literature. A contributor to AAC since 1982, he has authored over 230 articles and reviews on audio technology, music, and recordings. He has been the Sound Recording Reviews Editor of the ARSC *Journal* (Association for Recorded Sound Collections) since 1995, was co-chair of the ARSC Technical Committee from 1996 to 2004, and has given numerous presentations at ARSC conferences (www.arsc-audio.org). Mr. Galo is also a frequent book reviewer for *Notes: Quarterly Journal of the Music Library Association,* has written for the *Newsletter of the Wilhelm Furtwängler Society of America,* and is the author of the "Loudspeaker" entry in *The Encyclopedia of Recorded Sound in the United States,* 1st edition.

Chuck Hansen (Tascam CD-RW900SL–Measurements, p.33) is an electrical engineer and holds five patents in his field of engineering. He began building vacuum-tube audio equipment in college. He plays jazz guitar and enjoys modifying guitar amplifiers and effects to reduce noise and distortion, as well as building and restoring audio test equipment. He enjoys sailing and has over 150 magazine articles to his credit.

John Shand ("Jazz Track," p. 46) is a CD reviewer for *Australian HiFi and Home Theatre Technology.*

even enable a completely trashed disk to play and not sound too bad—which might be worth it. As such, any legitimate claims should be based on that possibility, and any tests should be based on those claims.

It is quite simple with a computer to "rip" a CD—that is, to extract the numbers exactly as they appear on the disk. There are databases available on the Internet which can be used to verify that the numbers read are correct, and disks can be easily compared.

Unfortunately for these products, if the numbers are correct to begin with, then the disk is "a perfect read" and there is nothing that can be done to improve it—by any tweak product or super super drive (in fact, any change would mean that errors were added). If, for example, you rip a disk, then treat it with "marvelous wonder cream," and again rip it—and the numbers are the same—then the cream didn't change the way it sounds, *period*.

(If the disk had errors, they were entirely removed by the digital error correction in both instances, and it will sound the same. It is possible that the cream could remove some errors, which might make certain drives more likely to be able to get a "better read" on a damaged disk—but it cannot help an undamaged disk. There are special pieces of test equipment which can measure the errors before they're corrected, which could be used to perform this type of test—but most drives don't report on corrected errors.)

If you rip a disk, and the AccurateRip database says it was perfect, and then you treat it, and it's different when you rip it the second time, then the product changed it—which means that the product introduced errors.

Incidentally, computer disks, unlike audio disks, lack that last "interpolation" level of correction (because data that is "almost right" is still no good). The fact that Mystery Cream enabled the author's computer disk to read after the treatment means that it *did* repair some of the surface damage (it either polished out the scratches or filled them in with something optically similar to the plastic of the CD/DVD)—at least enough to allow the drive to read the disk correctly afterwards.

This means that the product is actually useful for repairing damaged disks, but their claims are bogus—their product probably will make damaged disks sound better (or even perfect), but cannot possibly improve the sound of an undamaged or only slightly damaged one. In fact, it sounds useful for computer disks—but, in that market, they're competing against several relatively cheap alternatives. (I suspect that their marketing department has figured out that very few people will buy something expensive to treat the occasional damaged disk, but lots of people will buy something that supposedly makes all their disks sound better.)

For anyone interested in playing around with this stuff, the "best" ripping program is *free*—it's called EAC (ExactAudioCopy), and can be downloaded from www.exactaudiocopy.de. The EAC program is completely trustworthy and contains no adware, but you need to sort of dig the download link out of the other junk on the site lately (you'll be wanting EAC_0.99pb4.EXE or something like that).

A nicer—but not free—ripping program is called dBPowerAmp (there is always a current link to their site from the AccurateRip site).

EAC and dBPowerAmp link up with AccurateRIP (you can follow the info on the EAC site to AccurateRip for the latest info and a better explanation). AccurateRip does two things. First, it allows you to "calibrate your CD drive." This has to do with a certain timing "error" inherent in the way that CD drives read audio discs which makes the first few bytes they read invalid—AccurateRip asks you to read a known disk on your drive and calculates and verifies this correction number for your drive. Second, once your drive is calibrated, every time you rip a disk AccurateRip compares a checksum of your rip against a huge database, and informs you whether you have ripped a perfect copy—which is *very* handy.

I hope I've clarified things a bit.

Keith Levkoff
kLevkoff@panix.com
aX

Jazz Track By John Shand

www.audaud.com

Steve Hunter Band|Dig My Garden|Birdland BL015

Steve Hunter has long been alive to the possibilities of cross-pollinating flamenco with jazz. Its intrinsic drama plays to the strengths of Hunter's potent band, and this studio album's very live sound captures the visceral excitement. *Cazador* is the band's finest recorded work to date. It begins with scintillating flamenco bass from Hunter over disquieting, distant organ from Matt McMahon, who switches to piano to share with Hunter a pensive theme that still sustains the suspense. When drummer James Hauptmann gatecrashes this reverie with a thumping groove, the stage is set for James Muller to unleash a guitar solo of such ferocity you will be ducking for cover; the sort of playing that makes his live performances so extraordinary, but which often has been somewhat pasteurized on disc. Not this time.

The catholics|Village|Bugle BUG 007

Last year The catholics appeared to have reinvented themselves with the release of "Gondola." The relentless chirpiness was replaced with a broader emotional palette, painted on sparser, more atmospheric canvases. That reinvention was apparently short-lived, however, because with Village they return to the breezy rhythms and perky melodies that made the band so popular. I prefer the "Gondola" mood, here preserved for one track via the serenity of slide guitarist Bruce Reid's *Grace.* Nonetheless there is much to satisfy amid the jollity, as when Sandy Evans' soprano saxophone suddenly takes glorious flight from the fluid Afro groove of guitarist Jonathan Pease's *Pelicans.* Leader Lloyd Swanton has only penned two tunes this time, the repertoire being fattened with Bernie McGann's much-loved *Salaam* and Barney McCall's insistent *Ransome.*

Adrian Klumpes|Be Still|Leaf Bay 56CD

One of the moods music does best is eerie, and it doesn't come much more eerie than this. Adrian Klumpes, having made his mark in the nu-jazz of Triosk, has excelled on his solo debut. The sounds, all from a piano, are twisted by mechanical preparation or tortured by electronic treatment. Even the "straight" piano of the bewitching title track has some subtle treatment, so the notes seem to drift in a disquieting limbo. Klumpes' concept is not purely about texture and effect, having at its core a very pure beauty, charged with Gothic expressionism. Radical sounds are controlled with the same finesse as the keyboard itself, the notes sometimes shattering like fine crystal over ominous rumbling. The nursery-rhyme sparseness of the melodies and the use of repetition compound the pervading unease.

John Coltrane|My Favorite Things: At Newport|Impulse! 0602517350540

The only unreleased material here is a 23-minute *Impressions,* and even that has been available in edited form on "To the Beat of a Different Drum." The remaining five tracks, recorded at the Newport Festival in '63 and '65, appeared on either "Selflessness" or "New Thing at Newport." So does less than one track justify the CD's price for avid collectors? Probably. The '63 set had Roy Haynes rather than Elvin Jones on drums, and it's fascinating to hear his lighter but more bustling approach in the middle of the Coltrane/McCoy Tyner/Jimmy Garrison whirlpool. He's generally more reactive than the proactive, climactic Jones, although when Tyner and Garrison drop out after their brilliant (previously deleted) solos on *Impressions,* they leave Haynes and Coltrane igniting each other in an 11-minute tenor/drums inferno.

Blow|Blue Sun Red Moon|Birdland BL014

Drummer Ted Vining and pianist Bob Sedergreen have never left any doubt that their music has heart. Vining has often literally shouted it, with rambunctious cymbal crashes and reckless punctuations. These two Melbourne stalwarts continue to galvanize other players with their energy and sheer joy in making music. Alto saxophonist Peter Harper stands out, rushing to center-stage in the opening *And Heaven Called* with a sound as electrifying as a police siren in a drug cabal. It's broad and braying, and an ample match for any boisterousness Vining can hurl at him. The band members, completed by trumpeter Ian Dixon, trombonist Simon Kent, and bassist Mick Meagher, have penned the tunes. As the sextet's name suggests, they are open-ended blowing vehicles, and the playing is guaranteed to raise the spirits.

John Zorn, Mark Ribot|Asmodeus|Tzadik/Birdland TZ 7362

John Zorn may be the best-documented composer and improviser ever. Here guitarist Marc Ribot teams up with bassist Trevor Dunn and drummer G. Calvin Weston to turn ten of John Zorn's compositions from "Masada Book 2: The Book of Angels" into searing rock extravaganzas. However stylistically diverse Ribot has been across his previous projects, nothing will have prepared you for this. Zorn's serpentine tunes are relentlessly blasted and twisted by the guitarist, and the presence of Weston—who famously fuelled some of Ornette Coleman's harmolodic adventures—hugely compounds the ferocity. Imagine a blend of James Blood Ulmer and John McLaughlin at his most savage, on, say, Miles Davis's *Tribute to Jack Johnson,* and you have a sense of the cyclonic force; an overkill that would be dangerous, were it not such bone-jarring fun.