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AD STANDARDS

The UK's Advertising Standards Authority exists to protect consumers from the lies and deceit that so many companies try to get away with in false advertising, something the audio industry is all too familiar with. The latest company to be put on notice for shady advertising is Russ Andrews Accessories that sells a multitude of audio equipment including speaker wire, power cables, and interconnects in the UK.

According to the ASA, Russ Andrews Accessories was challenged by a consumer on the following claims that showed up in their catalog.

1. "The key to success of our PowerKords is KIMBER's unique cable weave which has proven to dramatically reduce Radio Frequency Interference (RFI) already on the mains supply and to reject further pick up of RFI ...," because he believed the PowerKord cable would have little effect on conducted electromagnetic interference;
2. "... Distortion levels inside equipment is vastly reduced, letting you hear a sound that is vastly clearer and purer, more detailed and far more dynamic ...," because he believed the Signature PowerKord cable would have little effect on measurable distortion in hi-fi equipment, and
3. "... eliminate system sound fluctuation and help to create a super-quiet noise floor, allowing more believable dynamics, deeper bass and lower high frequency distortion ... Listen out for a quieter noise floor (expect more dynamic music and greater detail) and a much more cohesive musical sound ...," because he believed the advertised spike-protecting devices would have little effect on the noise floor in hi-fi equipment.

So what does the ASA do? Submit the claims to an expert for verification, of course. The expert determined that the claims were in fact unsubstantiated and that the research papers provided offered no supporting measurements and were not peer reviewed to give them any credibility.

In the end, the ASA told the advertiser to cut the bullshit and, "not use the claims again unless they could substantiate them with robust scientific evidence." Ouch.

Now how do we get an agency like that over here in the US?

Source: UK Advertising Standards Authority

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Edward T. Dell, Jr. responds:

My longtime valued friend Dick Campbell is a highly talented engineer and teacher. His dedication to honesty and integrity where audio gear performance is concerned is second to none. Nothing arouses his righteous ire more than fanciful claims, unsupported by objective measurements, about wire especially. However, the brush he chose in his interesting letter about a British wire advertiser seems rather too broad. His instance cited is not a group of advertisers, although the claims are too often typical of what manufacturers' ad agencies include in their ads.

I think it is important to realize that almost all advertising of anything is now based on the consumer's feeling. Very few ads for cars extol measured technical details. They emphasize how owning the expensive vehicle makes you feel when you drive it, are seen in it, break the speed limit in it. I suspect wire buyers who choose gold-plated #10 gauge for connecting car speakers to 1kW amps are not thinking of measurements but what their girlfriend will think.

So, Richard, I think your letter is well taken, but most people know that hyperbole is a constant hazard. And it is fair to remember that not all cable ads are guilty—especially not by association.

AMP6 AMP

J. R. Laughlin's description of his AMP6 amplifier (March '08, p. 18) is solidly detailed as to layout and construction, but somewhat sketchy as to circuit operation.

The driver/phase inverter (6SL7) belongs to the class of so-called "floating" paraphase inverters. In particular, the author uses the "see-saw" configuration of M. G. Scroggie (*Wireless World*, July 1945). This and related circuits are treated in detail, with references, in F. Langford-Smith's *Radiotron Designer's Handbook*, 4th edition.

Mr. Laughlin's comment that section U2b of the 6SL7 is configured in a manner akin to that of a unity-gain inverting amplifier is well-taken and certainly not described in those terms by Langford-Smith. In fact, the term "floating," which refers to the voltage at the junction of R3 and R9 in Figure 7, would today be called a "virtual ground." If R3 = R9, then the junction of these resistors would approach zero AC volts to ground as the gain of U2b increases. (Because there is no DC current in these resistors, the DC voltage to ground is immaterial.) If U2b had the gain of a typical IC amplifier, there would be no need to adjust the ratio of the resistors for equal (but opposite) drives to the 6V6 grids. Using the equations given in the *Handbook* and assuming a gain of 40 for U2b, Mr. Laughlin's selections of 470K for R3 and R14 and 510K for R9 should result in drives balanced to about 1% or better and, of course, he allows for the use of a potentiometer to adjust for even closer balance.

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MAINS ANALYZER

I would like to compliment Iain McNeill on the excellent construction article ("A Mains Analyzer," Nov. '07, p. 8)—and an interesting choice of what to build. Personally, I never doubted exactly how bad the AC line was, but it is interesting to see that fact quantified.

There are, however, one or two of Iain's conclusions with which I disagree (and one or two I would add).

In a digital switching amplifier, contrary to what Iain suggests, the output tracks the input due to feedback, and the power supply is merely used to supply "raw packets of current" to what is essentially a reservoir—again under the control of feedback. All of the power is used in the form of high-frequency pulses, which are quite easily filtered and not terribly sensitive to individual variations. Most digital amplifier designs are, therefore, quite immune to supply variation or noise (luckily). [This may not be true of some older designs that operate as "linear converters" at relatively low frequencies.]

To me, the entire issue simply stresses that care must be taken when designing a power supply. With a power supply, you are basically starting with a 100% AC noise component, which is converted to (you hope) clean DC by the supply. Iain's findings stress the need for good power supply design, good circuit choice, and good matching between the two.

Solid-state amps should be designed with symmetrical supplies (noise which appears equally and out of phase on the plus and minus supplies is easier to reject). They should use sufficiently large filter capacitors and decent high-frequency bypass capacitors to remove the noise. Many poor designs scrimp on the filtering and rely on the PSRR of the circuitry alone—which should be avoided unless the PSRR is very high (and even then on general principles). Taking care to ensure that the supply bridges are connected symmetrically, or using a 1µF polypropylene bypass cap instead of a 0.01 mica, can make a big difference in supply noise, but often manufacturers or designers don't bother.

Sensitive circuits, particularly those with poor or no PSRR, should be supplied with regulated power—in both tube and solid-state equipment. If you're going to build that SET amplifier, you probably *should* regulate the output stage power supply. Yes, this will make the job harder, or it just might suggest that a single-ended amp isn't such a good idea after all. Your basic push-pull amplifier uses the output transformer to cancel out the noise on the output stage—so is a lot less fussy about supply noise—sounds like a better design to me. The driver stages, not being push-pull, should be given very well-filtered power or, better yet, regulated power. Because they don't draw much current, this isn't a hardship—and even an R-C-R-C-R-C filter will do a lot of noise suppression.

I would especially recommend Iain's article to anyone even considering purchasing one of those expensive line cords to attach his (or her) equipment to the mains. Ditto for fancy wall outlets. The very idea that anything is going to influence the way something sounds by somehow doing a better or worse job of passing on the crud that's coming out of the wall is pretty laughable (beyond a decent, thick, low impedance piece of copper to avoid voltage drop).

[If you're using a good power conditioner, the power going into it is the same crud from the wall, so a fancy cord or outlet won't help; the power it's sending out to your equipment should be pretty good already, so it doesn't *need* any help from a fancy tweak cord either.]

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Iain McNeill responds:

Thank you for those kind words and for your interest in my article. I took a look at the current offerings from TI, ADI, and Tripath and found that you are quite correct. The latest consumer-audio targeted products do have feedback loops and therefore can be expected to offer some degree of power supply rejection. My experience with class D was some time ago with custom chipsets for the wireless communications industry and had no such feedback; you could hear exactly what the power rails were doing. As always, buyer beware!

I completely agree with your comments regarding power supply design for audio. The biggest problem with power supply filtering is that the source impedance is so low, making it difficult to get significant attenuation. A typical 40W/channel stereo amplifier will have a transformer secondary resistance in the order of 0.3Ω (for 10% regulation), and so even the best electrolytics, with series resistances in the 25mΩ range, will give only around 20dB attenuation. Paralleling multiple capacitors helps, but is subject to the law of diminishing returns.

As you point out, it is important to use multiple types of caps because large electrolytic capacitors tend to have large parasitic inductance which increases their series impedance at high frequency. Personally, I would use the poly 1µF *and* the 0.01µF mica on top of the electrolytic reservoir caps and probably a 10pF as well for good measure. Using two of each type in both positive and negative rails would not be paranoid in my books! This is in addition to the liberal sprinkling of decoupling caps around the circuit. Obviously, if you can afford a little resistance in the supply lines, then things improve quickly and the R-C technique, as you describe, is very effective at isolating sensitive input stages from noisy supplies as well as each other.

I think the jury is still out on those fancy power cords. In some ways it is easier to filter on the line side of the power supply. On the 120V side you can have 9Ω of source resistance and still get 10% regulation for that 2 × 40W amp; now that capacitor is giving you 50dB of attenuation. Throw in a ferrite sheath (as some vendors claim) and you could potentially have some pretty decent broadband attenuation, albeit at the expense of some extra ground leakage current and some possibly significant heat dissipation. But as you say, this is what the power conditioner is doing anyway. I don't have any data to support any of this, but, theoretically, the approach does seem to have some validity.

OLD-TIME AMPLIFIER

I read the article by Mr. Brown ("Upgrading a Classic Pilot Amp," April '08, p. 8) with interest—very nicely done. The single grounding bus is how it was done in the 1940s and 50s, and how it should be done. One comment would be that lower hum can be achieved by mounting the mains transformer so that the windings are orthogonal to the plane of the chassis. It seems to have been done that way in the unit that Mr. Brown described. In addition to mounting the windings orthogonally to the chassis, I obtained additional hum reduction by connecting the mains transformer to the chassis in a single point.

I have studied the noise and fidelity of old amplifiers and discovered that electrolytic capacitors of old vintage have such a high and noisy leakage current that much rumble and "cooking" sound can result from those capacitors no matter where in the circuit they are used. It is a good high fidelity upgrade to replace those capacitors with a modern construction. Modern electrolytic capacitors are orders of magnitude better than old vintage ones. The manufacturers say that the old capacitors were not made of pure enough materials and will be noisy forever, if they last that long.

Cathode bypass caps should be of the modern variety as well. What I have found to improve the fidelity of an amplifier the most, however, is to use modern film capacitors for interstage coupling. Old paper and wax or paper and oil capacitors have significant leakage currents, which are very good low-frequency noise generators. An amplifier built with those capacitors will sound like someone is cooking water in a big pot when you place your ear to the speaker. You often get that same sound when you "recycle" old tube-sockets, especially the flat bakelite ones from cheap radios.

A trick we used in the 50s was to use carbon comp resistors of about five times the dissipation capability as the power-dissipation of the circuit itself. If, for instance, a resistor dissipated 0.5W, we would use a 2W resistor there.

From my experience, you can replace rectifier tubes with solid-state rectifier diodes as long as you place 2,200pF, 2kV capacitors across the diodes and a 10W 470Ω resistor in series with each diode. This circuit mimics the performance of a rectifier "bottle" so well that I doubt that anyone can tell the difference.

By all means put in a rectifier to make it look like you are doing the "right thing," but do not connect the rectifier heater to B+. Old mains transformers do not like the 300 or more volts on the rectifier heater for very long. So much moisture has become absorbed by the insulation over time that the electrolysis will destroy the transformer in a few months. At least that has been my experience.

There may be some cathode stripping from applying B+ before the cathodes in the rest of the circuit are hot, but that happened with 5U4 directly heated rectifier tubes as well, and I never experienced any problems with cathode stripping in my designs whether I used solid-state rectifiers or not.

Hans J Weedon
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SPEAKER PLACEMENT

Like many things in audio, Ambiophonic things are relative. Ambiophonics and RACE crosstalk cancellation—both in theory and in practice—work best with speakers that are relatively close together compared to the stereo triangle. Speaker angles from, say, 15 to 30° are optimum in my experience. There is no one value that is the golden angle because speakers, audiophile preferences or head sizes, and recordings vary.

It is easy, however, to find the optimum angle for your speakers and ears. Simply listen to just a left input signal and adjust RACE and the speaker angle until the signal is as far to the left as you want it to be. I like it at the extreme 90° position, which is where the theory says it should be if a signal is all left. But this is difficult to obtain for this particular test signal that has no interaural time difference, only the level difference. Then listen to an all-right signal and make sure things are the same for this case.

You will find that if the speakers are at the normal stereo angle of 60° apart, then the stage will not be very much wider than that but it still does something and it is better than stereo for most listeners. But the wide angle lessens the pinna advantage of the narrower angle and the sweet area is usually reduced, room response is harder to control, and so on. If you are an audiophile or appreciate high fidelity, I can't imagine you will be happy with compromised Ambiophonics.

In Roger Russell's reply (Jan. '08, p. 49) he writes about speaker spacing in feet, and so on. It is not the lineal distance between the speakers that is important; it is only the angle they subtend to the listener that is significant. I don't know about the SRS system, but in Ambiophonics as you move back along the center line away from the speakers, the width of the stage does not change very much. Normally, you cannot get far enough away from the speakers in the average room to lose the stage. If you get too close to the speakers, you begin to hear ordinary stereo. I have had six to ten people in a sloppy line all hearing a train go across the room.

Most of the adjustments in RACE (both in the TACT box and in the free software version), if not radically altered, are really inaudible to people with normal hearing. They are there to cope with unusual situations, but may be useful to audiophiles who can hear differences in cables and such, and are skilled in tweaking.

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Roger Russell responds:

I hope, Mr. Glasgal, you were not offended by my example of listening at a 10′ distance. However, 10′ can be a typical listening distance for many people at home. Of course, speaker placement, listening position, and so on can be adjusted, but that was not the point of my reply. Although your reply has provided another opportunity for a sales presentation for your system, it in no way invalidates my example given in my previous reply.

You may find that crosstalk correction using my column speaker systems described in the November 2005 and July 2006 issues of *audioXpress* can provide better coverage. All of the drivers are in the same vertical line and each covers the entire frequency range. As a result, the excellent coherence of the sound can enhance the effectiveness of crosstalk correction. In addition, the sound will appear the same regardless of your listening height, and the columns project an expanded near-field.

Of course, in the final analysis, many "stereo" recordings are not made with only two microphones and we are limited in what we hear by various recording methods of miking, mixing, panning, and so on. Crosstalk correction is not a method of reconstructing what we might hear by attending a live concert. Instead, the correction adjustments can yield pleasing effects with many alternate realities, none of which may duplicate the original recording.

The best that can be done to eliminate stereo crosstalk is to use headphones, and that eliminates listening position, room effects, speaker directional characteristics, and so on. However, I think that most of us don't want to wear them. Perhaps binaural recordings could at least offer an improvement when played with crosstalk correction through two speakers.

CORRECTION

I have questions for Paul Loewenstein

on his article "A Dynamic Microphone Soundcard Amplifier (Feb.'08, p. 22)."

1. Under "Open Loop Gain": "3.5 – 1.2 = 1.3," not 2.3.

2. Then "gain is about 40 × 1.3 = 52." Where does the 40 come from? Perhaps it should be 40 × 2.3 = 92.

I suggest this calculation:

$$r_e = \frac{26\Omega}{I_e}\text{, where } I_e \text{ is in mA}$$

$$I_e \cong I_c = \frac{\text{Ring} - V_{B2}}{R_2} = \frac{3.5 - 1.2}{5.6\text{k}} = 0.41\text{mA}$$

$$r_e = \frac{26\Omega}{0.41} = 63\Omega$$

$$A = G_{Q1} \cong \frac{R_2}{r_e} = \frac{\text{Subk}}{63\Omega} = 89 \text{ (close to 92)}$$

3. "Z_{in} = 29kΩ." Another way is: $Z_{in} = B_{re}$ = 270.63 = 17kΩ.

4. Closed loop gain equation is wrong. "Gain of 19." Analyzing Fig. 2, I get:

$$\frac{E_o}{E_i} = \frac{1}{\frac{1}{-A} - \frac{x}{-Az} - \frac{x}{-Ay} + \frac{x}{y}} \cong \frac{1}{\frac{1}{-A} + \frac{x}{y}} = \frac{\frac{y}{x}}{1 - \frac{y}{Ax}}$$

= 45, which makes sense. If $A \rightarrow \infty$

then gain = $\frac{\text{feedback}}{\text{input z}}$ as with an op amp, which would have a gain of: gain =

$$\frac{10\text{k}}{323} = 31.$$

So this circuit has less feedback and more gain.

I like this circuit. Straightforward.

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Paul Loewenstein responds:

Wow! The first version of this article was circulated to a speech recognition user group in April 2003, was published before in *audioXpress* in September 2005, and you are the first to notice this elementary arithmetic error. Yes, 3.5 - 1.2 = 2.3. The open-loop gain calculation of 40 times the voltage across R2 corresponds to:

$$r_e = \frac{25\Omega}{I_e}$$

(with I_e in mA), which corresponds to a temperature of 290K rather than 300K. This makes little difference once the arithmetic error is corrected.

It would probably have been clearer had I gone back to first principles and calculated the grounded emitter gain and input impedance using your approach. However, once the arithmetic error is corrected and the temperature adjusted, the two approaches give identical results.

The error in the closed-loop gain equation is an error introduced in typesetting. The submitted equation (and the one published in September 2005) is:

$$\frac{-Ayz}{xy + yz + (1 + A)zx}.$$

The typeset version needs parentheses around the denominator because it is laid out as a single line fraction. I rechecked the submitted equation and I believe it to be correct. Your calculation of closed-loop gain appears to be incorrect; with the limited open-loop gain, the closed-loop gain should be less than the $A \rightarrow \infty$ gain of about 30.75.

My arithmetic error propagates to my understating open- and closed-loop gains and overstating harmonic distortion. I provide corrected figures here:

Open-loop gain 92
Grounded-emitter input impedance 16.4k
Closed-loop gain 23.2
Variation of grounded-emitter gain ±1%
Variation of emitter-follower gain ±0.4%
Variation of open-loop gain ±0.6%
Variation of closed-loop gain ±0.15%
Harmonic distortion 0.04%

The variation of gain and harmonic distortion assume ±1mV peak signal from an open-circuit microphone, giving ±23mV peak output. Harmonic distortion is directly proportional to the signal level.

Your correction has reduced the calculated harmonic distortion by a factor of 4, via a combination of reduced open-loop harmonic distortion and higher open-loop gain.

I can also take this opportunity to point out that the version published in September 2005 overstates harmonic distortion by a factor of 2 relative to the already overestimated gain variation. Thank you for your diligence in picking up this error. *aX*