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CORRECTION

Equation 4 in my article "A Flexible Subwoofer Amplifier" (Jan. '07, p. 37) contains a couple of typos. The correct equation should read:

$$\text{Phase (rad)} = \tan^{-1}\left[\frac{\frac{2\omega}{R_1C_1}}{\omega^2 - \left[\frac{1}{R_1C_1}\right]^2}\right]$$

Rudy Godmaire
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FLEXIBLE SUBWOOFER

I enjoyed Rudy Godmaire's article in the January issue, but have a few comments. There are several instances in which polarized capacitor symbols are used, but I'm sure Mr. Godmaire intended to show film capacitors:
Figure 2: C29, C31
Figure 6: C3, C4, C5
Figure 10: C17, C21

There is no parts list in the article to verify the capacitor types he used and, unfortunately, tantalum polarized capacitors are available in some of these values. They would be most unsuitable for the filter frequency determining capacitors.

I am concerned with his method of powering LED D6 through the two series 24V zener diodes ZD5 and ZD6 without any series resistance. The dynamic impedance of a 24V 5W zener (such as the 1N5359) is only 3.5Ω. With 5% tolerance zeners, the breakdown voltage could be as low as 45.6V. Any surge in the +48 supply could deliver high damaging current to the LED.

Another concern is the use of pots P4 and P5 to set the DC rail voltages for the op amps (Fig. 11). The most likely failure mode for any potentiometer is an open wiper. When (not if) this occurs, one of the V3 voltages would increase from 8V to 13.8V. It is much better in terms of noise and reliability to use 1% fixed metal film resistors to set the rail voltages. Op amps are relatively immune to small differences in the rail voltages, but they may not appreciate a 73% difference.

I'm not certain why he used a supply voltage of only ±8V for the OPA-2604 op amps, and he didn't elaborate on his choice in the article. The OPA-2604 has a maximum supply limit of ±24V, and is safely usable to ±21V. I've seen articles that state some op amps sound their best when used at higher rail voltages, but this may be op amp specific. I don't think there is any downside to using the traditional ±15V rail voltages. He could raise the input voltage to the linear regulators to ±18V, there would be less power dissipated in IC4 and IC5, and he could reduce the dissipation in the series zener voltage string ZD1 to ZD4. The remaining dissipation resulting from stepping the original ±48V down to ±15V would be spread across all the op amps, which are well below their maximum rated dissipation.

While Mr. Godmaire had a space limitation in his amplifier, if the reader has more room, he/she could use a small separate 25.2 VCT "filament" transformer to provide the DC input to IC4 and IC5. This would eliminate the power dissipation in the zener diodes and provide a lower voltage for the LED indicator series resistor. The entire amplifier would benefit from the reduced power dissipation.

Finally, I am firmly in the camp of Gary Galo when it comes to adding protection diodes to linear regulator ICs (see "Preamp Power Supply," *TAA* 4/90, p. 47). Mr. Godmaire did not violate any by-the-book rules in his linear regulator supply application, but five more 5-cent 1N4002 protection diodes are cheap insurance, and as Gary says, their use is not negotiable.

This is a well thought-out article, and I certainly look forward to Part 2 of Mr. Godmaire's article.

Chuck Hansen
Orange, N.J.

Rudy Godmaire responds:
I would like to thank Mr. Hansen for his interest in my article and for taking the time to share his concerns. I feel privileged to receive comments from such an authority.

Regarding Mr. Hansen's first concern about the potential misuse of polarized capacitor symbols, in Fig. 2 C29/C31 are film capacitors and the symbols appearing in the schematic seem to be appropriate. As I mentioned at the end of the article, all op amps have been bypassed with 10µF/16V electrolytic capacitors and 0.01µF/63V polypropylene capacitors (pins 4 and 8 in the schematics). The parts lists will appear in Parts 2 and 3 as support information for the construction of the amplifier.

C3, C4, and C5 are electrolytic capacitors. I assumed the circuit could allow for the use of these types of shunt capacitors. Measurements in Part 4 tend to confirm that. Having said that, there is no doubt Mr. Hansen's suggestion will improve the performance of the filter, and I will take the opportunity to replace these electrolytic capacitors (Black Gate) with metallized film capacitors.

For now, readers may already consider replacing the parts shown in Table 1 of Part 2 with the following metalized film capacitors:

C3: 0.22µF/50V	Digi-Key Part No.: P4667-ND
C4: 0.33µF/50V	Digi-Key Part No.: P4669-ND
C5: 0.47µF/50V	Digi-Key Part No.: P4671-ND

As a better alternative, you could replace C3 and C4 with a single 0.56µF/50V capacitor (Digi-Key Part No.: P4672-ND). Note that I chose this model of metalized film capacitors mainly for its small size. Other brands or models will do.

While Fig. 10 shows C17 and C21 as electrolytic capacitors, they are, in fact, polypropylene capacitors. This detail escaped my attention while proofreading the article, and I thank Mr. Hansen for pointing out this mistake.

Though Mr. Hansen is concerned with my method of powering LED D6, I can report that I have never had any problems so far (two years). If I may build on his suggestion, LED D6 might simply be connected through a series resistor to V3+, without the need for any zener diode.

I am surprised to read that Mr. Hansen is concerned about pots P4 and P5. The voltage regulator datasheets of Linear Technology suggest, in many schematics, the use of a potentiometer as a means to adjust the output voltage. I have selected quality cermet potentiometers for this task and I am confident they will provide reliable operation for years to come. Nevertheless, Mr. Hansen's intervention is certainly guided by his longstanding experience and expertise. I will retain his advice for future projects.

Like Mr. Hansen, I also heard about the benefits of raising the supply voltage of op amps. Unfortunately, this was not the case in my application. The supply voltage of ±8V provided me with the lowest level of hum and noise (H&N). As mentioned in Part 2, I measured 1.0mV of H&N at the binding posts with the volume control at full level.

As an example, increasing the supply voltage of the op amps to ±10V increased the H&N to 1.7V. Further increase of the supply voltage would worsen the situation. Not being an expert, I can only speculate on the reasons for this situation. My best guess is that it might be related to the design of my PCBs as discussed in Part 2.

Finally, I am glad to know I did not violate any by-the-book rules in my linear regulator supply application. The circuit was designed with full respect to Linear Technology's recommendations described in their datasheets. Though I did not read the article Mr. Galo published some time ago, I have already prepared a revised version of PCB# 4 that takes his sound advice into consideration. Readers may communicate with me through my website (www3.sympatico.ca/r.godmaire) should they wish to get a copy of the revised templates.

I hope Mr. Hansen will enjoy the rest of the article. I invite him to keep on expressing his concerns, or appreciation, for the benefit of all readers, including myself.

MUST-SEE SEMINARS

I just finished reading the article "Grounding and System Interfacing" by Gary Galo in the January 2007 issue of *audioXpress*. It was a great distillation of many of the most important factors that affect noise in sound systems. I was especially interested to see the new balanced ICs available from THAT Corporation. Nice job, Gary!

In 1999, the Chicago section of the AES hosted a seminar by Neil Muncy titled "Grounding, Shielding, and Things that Go ZAP! in Audio Signal Processing Systems." I was fortunate to attend, and to say it was an eye—and ear—opener is an understatement. Through a series of visuals and live demonstrations, Muncy led the attendees through the various aspects of the power distribution system and also the correct and incorrect ways to handle noise suppression. At one point in the demo, he showed how effective proper techniques could be by running a line level signal about 200′ through the room including over fluorescent lights, video monitors, and other noise-generating devices with virtually no noise, even though the various pieces of equipment in the test were powered from different lines within the building. Truly amazing!

The one thing I did not see mentioned in Gary's article was the pin 1 problem. Even though devices may be connected using balanced lines, if the termination of the shielding connected to pin 1 is not done properly, noise can creep in. Connection of pin 1 to the chassis as close to the entry point as physically possible is the proper choice. A bad choice is to connect pin 1 to signal common even if it ultimately is connected to the chassis. I hope by now most commercial equipment will be connected properly, but if you are doing your own projects, this is important to remember.

If any reader ever gets the chance to attend seminars by either Mr. Whitlock or Mr. Muncy, they should run—not walk—as fast as they can for an experience not to be forgotten and immensely valuable.

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Gary Galo responds:

Mr. Perazella is correct in pointing out the "Pin 1 Problem" (a term coined by Neil Muncy). Bill Whitlock did discuss this problem in his presentation, and readers should refer to the .PDF documents that I mentioned on pages 29 and 30 of *aX* 1/07 for detailed information on this issue. Essentially, pin 1 problems occur when star grounding has not been observed in equipment design and layout. On page 120 of Whitlock's Power Point presentation, he illustrates both incorrect and correct methods for connecting pin 1 on XLR input and output connectors inside the equipment chassis. Incorrect methods include various daisy-chain ground schemes and others that result in multiple paths back to chassis and/or power supply ground. Correct methods bring all XLR input and output grounds to a single "star" chassis ground near the power supply.

Mr. Perazella is correct: No one who works in audio should miss the opportunity to attend a presentation by Bill Whitlock or Neil Muncy.

HUM PROBLEM

In constructing amplifier circuits—namely, preamps—the most frustrating thing is hum. A poor design, or perhaps a simple construction error, can result in a circuit that hums like a buzz saw.

The first thing you tend to blame is a faulty ground. If that's OK, then you suspect inadequate filtering in the power supply.

You eventually get to the point where you've checked every possible culprit—except for one, which is most often overlooked—the power transformer.

Because of the principles in which it functions, the power transformer behaves like an electromagnet whose field reverses 60 times per second, as does the power being applied to its primary. In turn, the chassis on which it is mounted becomes magnetized as well. The magnetized chassis won't attract a tack or nail—it's not that strong. It is, however, a strong enough field to be easily picked up by the preamp tube, usually a 12AX7.

This was proven when I disconnected the power supply of one of my "buzz saw" preamps, and obtained the heater and plate voltage from a second preamp, which also had a horrible hum. The result was a loud hiss—not a lick of hum whatsoever.

I've tried newer hi-tech power transformers and still had hum. I've also tried plastic spacers between the transformer and the chassis, and that didn't help. The preamp tube and the power transformer are like two guys, each from a different street gang that cause trouble when they come near each other. Separate the two, and the result is peace and quiet.

Through experimenting I've found that when using an outboard power supply (no flowing AC or magnetic fields near the preamp tube), even the simplest power supply will do just fine. No need to fool with exotic capacitors and regulators. With the power transformer away from the preamp, construction is no longer critical; it still sounds great.

I thought Joseph Norwood Still's preamp (*aX* 3/01 p. 36) would put a dent in my theory here. With the power transformers (three in all!) mounted on the preamp chassis, he reported no noise, hiss, hum, or whatever with the volume at ¾ rotation. Then I remembered that when a turntable is off, the cartridge is shorted out (as is the preamp input as well). Any preamp will be quiet when its input is shorted to ground. An externally powered preamp is quiet with the turntable connected and switched on.

I hope this helps solve at least one DIYer's hum problem.

Neal A. Haight
Castro Valley, Calif.

DOPPLER DISTORTION AGAIN

While perusing my *aX* library, I noticed a letter response by G.R. Koonce presenting a good case for the audibility of speaker Doppler (FM) distortion (*aX* Aug. '02, pp. 60-62). My enclosed *Fig. 1* shows the derivation of his equation. But I propose an additional factor, the increased audibility and objectionability with increasing LF modulating frequency (f_1), whereas the actual distortion sidebands are independent of f_1. Three examples show the rationale:

1. A singer's or instrument's vibrato (FM) rate sounds pleasant around 5-7Hz; faster rates sound "choppy" (as many inexperienced singers have yet to learn).
2. If a voltage-controlled oscillator (vco) is frequency modulated at rates above 20Hz, the sound is very discordant.
3. An extreme example is shown in the data row in the *Fig. 1* table with f_1 = 0.1Hz, A_1 = 0.5″ peak, and f_2 = 300Hz. Unweighted Doppler distortion is 4.9%. But consider that the speaker cone's f_1 takes five seconds to move forward, and five seconds to move back. Intuitively this would be no more audible (disregarding motor/suspension nonlinearity) than moving your head forward 1″ in ten seconds!

The reason is that, while there is 4.9% FM "distortion," the (predominant) 1st order sidebands (at –29dB) are only 0.1Hz away from the 300Hz input. And the peak frequency deviation Δf_{zp} is only 0.007Hz, a change of only 0.0004 semitones, which is certainly inaudible!

Thus I propose the factor "S" in *Fig. 1*, standing for "subjective Doppler distortion." "S" is a weighting factor equal to the ratio of f_1 to 20Hz. 20Hz is arbitrary; it's the proportionality to f_1 that's significant.

Of practical significance, note the example that Mr. Koonce presented (the first row in *Fig. 1*). As he said, the (unweighted) 33% Doppler distortion should certainly be audible. Paul Klipsch demonstrated the Doppler effect with an "unnamed" (the Bose 901) system using nine 4″ drivers. It should be easy to see on a spectrum analyzer, and you can rule out ordinary harmonic/IM distortion by choosing a very linear driver and viewing the 50Hz signal alone.

I agree with Mr. Koonce's observation of the superior midrange clarity (at high power) of a three-way versus two-way: I reviewed the excellent Swans M1 by itself (*SB* 3/99) and crossed over at 220Hz to an 8″ Scan-Speak 21W/8555-00 sealed sub (*aX* Sept. '05). With sub, it sounds clean even with my 125W per channel "Mad Katy" tube amp (*aX* June/July '06) at full power; the midrange was "muddy" before installing the sub. Likewise for my three-way "Venue" speaker (*aX* Nov. '06); very clean midrange. A notable two-way exception is Joe D'Appolito's "Thor" (May '02), which uses two 18cm (7″) mid/woofers with enough cone area plus TL loading to keep the bass excursion low enough to avoid audible Doppler (or any other) midrange distortion.

I'm sorry for being four years late in appreciating Mr. Koonce's letter, but I look forward to his thoughts on my thoughts regarding Mr. Doppler's thoughts!

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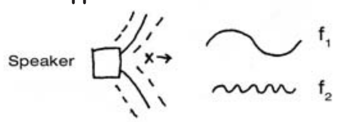
FIGURE 1: Doppler distortion



$x = A_1 \sin 2\pi f_1 t + A_2 \, \mathrm{Sin}\, 2\pi f_2 t$
Assume $A_2 << A_1$

c = speed of sound = 1128ft/second, M = modulation index = $\frac{\Delta f_2}{f_1}$

velocity $v_1 = 2\pi f_1 A_1 \cos 2\pi f_1 t$, peak velocity $(f_1) = v_{1p} = 2\pi f_1 A_1 f_2$ peak frequency deviation =

$$\Delta f_{2p} = f_2 \frac{v_{1p}}{c}, m = \frac{\Delta f_{2p}}{f_1} = \frac{f_2}{f_1}\frac{v_{1p}}{c}$$

$M = 2\pi A_1 \frac{f_2}{c}$; for $M << 1$, d =

$$\frac{\text{rmssidebandamplitude}}{\text{rmsun mod amplitude}} = \frac{m}{\sqrt{2}}$$

$d = \sqrt{2}\pi A_1 \frac{f_2}{c}$ = RMS Doppler distortion (upper and lower 1st order sidebands)

D = % Doppler distortion = $0.03282 A_1 f_2$ (A_1 is in peak inches). Define S = Subjective % Doppler distortion, proportional to $\frac{f_1}{20\text{Hz}}$;

$S = 0.001641 \, A_1 f_1 f_2$

HORN GUIDELINES

Ed Simon's article, "A Combination Horn You Can Build" (*audioXpress*, January 2007) was a real source of inspiration to me. I've always liked horn enclosures, and Ed's seems to me to be one that is thoughtfully designed, promises great performance, and is easy to build! What more could a speaker-putterer with marginal woodworking skills ask for? I really want to put together a pair of these, but feel a bit at-sea due to the lack of a dimensioned drawing. The Fig. 1 Cutting Pattern is a big help, and Photos 3 and 4 show the box in the process of assembly, but a picture's worth a thousand words and front/side drawing with all the hidden lines seems a good idea to make sure that it comes out right.

Jim Wood
Brea, Calif.

Ed Simon responds:

I am sorry I did not do a dimensioned assembly drawing when I built them. Because they are fully assembled and I do not have access to the interior, I cannot do an actual accurate as-built diagram. Perhaps someone else building them would be kind enough to do one.

Of course, if you want to try it out on your own without risking any serious investment in time or materials, you could build a prototype out of drywall. It is cheap and you need only a knife and straight edge to score and snap it.

Sorry about that. *aX*

A1 excusion, peak	f_1	f_2	d RMS	S RMS	relative amplitude 1st order sidebands, each of 2
0.5″*	50Hz	2000Hz*	32.8%*	82.1%	-12.7dB
0.125″	40Hz	3000Hz	12.3%	24.6%	-21.2dB
0.5″	20Hz	300Hz	4.9%	4.9%	-29.2dB
0.5″	0.1Hz	300Hz	4.9%	0.025%	-29.2dB
0.125″	100Hz	10kHz	41%	205%	-10.7dB

* Example shown by Mr. Koonce

aX Online Exclusive

Modifying Mighty Mouse

By Bob McIntyre

The Little Amp That Could Gets a Minor Facelift

I'm an inveterate tinkerer. Give me an amplifier—even a brand-new one—and I'll tear into it in an attempt to extract that last little bit of performance. This cannibalism extends even to those projects I've built myself.

My little tube amp Mighty Mouse appeared in the April 2006 issue of *audioXpress*. I thought the design was reasonably finalized, but in the year or so I've lived (and tinkered) with it, the relatively minor changes I've made have resulted in a huge improvement in its performance. ■

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