CORRECTIONS

I just realized that I have made a very minor error in my article "Diffraction Doesn't Have To Be a Problem" (*aX* 6/05). I referenced the types of felt used as being "AES," when it should have been "SAE" for "Society of Automotive Engineers." I seem to repeat this mistake frequently when referring to the description. I suspect that this won't cause any real confusion, but I wanted to correct the error.

David Ralph

There are errors in the schematics of my article "Pentode Petite" (June '05 *aX*).

1. R1 in Figure 1 should be 150Ω, not 1k5 ohms.
2. The polarity of C15 in Figure 2 should be reversed to match the polarities of C16 through C18.
3. L1 is not listed in Table 2; it is a 10H 100mA choke (Hammond 158M), available from Antique Electronic Supply or Angela.
4. In Figure 1, the output transformer secondary leads should be connected G (green) to GY (green-yellow) and B (black) to BY (black-yellow), and not the way they are shown.
5. Photo 4 in the text should be Photo 3, while Photo 3 in the text should be Photo 4.

I regret these errors and hope that builders notice them, especially the polarity reversal of C15.

Tim Smith
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ANALYZER REFINEMENTS

I'm very satisfied with the 1PPM IM Distortion Analyzer designed and described by Richard Crawford (*aX* March 2004) that I have built. It is very useful, works fine, and is almost indispensable for everyone active in the field of audio electronics.

I also added a 3½-digit digital panel meter. But after some experience I found an analog meter more convenient. The minimum floor distortion/noise level of the instrument (with the generator output directly connected to the input of the measuring part) was slightly less than 0.00003%, or 0.3PPM, in the 20 and 19.2kHz mode. In the other two modes the noise floor level was slightly worse at about 0.00005%, or 0.5PPM. This is possibly the result of the greater distance between 800Hz and 20kHz/19.2kHz than between 800Hz and 9.6kHz/10.4kHz.

In using the instrument, however, I discovered a remarkable phenomenon. When using the 50kΩ input attenuator (P2) of the measuring part (while increasing the attenuation), the noise/distortion floor increased considerably. I found that most potentiometers introduce a lot of distortion in positions between minimum and maximum. I tried several brands and types.

It seems that even new potentiometers from well-known brands have this shortcoming. It is probably because of some diode effect in the contact resistance between the rotating contact and the resistance strip. I also found that this problem is worse at higher potentiometer resistance values and also worse at logarithmic types compared to models with a linear function. The much lower value (1kΩ) attenuator at the output of the generator part hardly showed this problem.

I eventually replaced P2 with a dual type (stereo pair) of 100kΩ potentiometers from the ALPS brand with both parts wired in parallel. This gave me the best results. But maybe even better types exist.

Paul J. Bruinsma
The Netherlands

Richard Crawford responds:

I thank Mr. Bruinsma for sharing his experiences and for his recommendations. I would like to reply to his comments on the input potentiometer and its distortion/noise. First, there can be a type of noise (thermal noise) introduced by the input potentiometer, particularly at midrange settings. Thermal noise will be present in any potentiometer, regardless of quality. The thermal noise should be small enough not to interfere with the 1PPM capabilities, but it can affect the "noise floor" of the instrument.

In the design phase I was aware of the possibility that the input potentiometer could produce distortion. I selected a 24mm diameter input potentiometer to reduce the risk. In my measurements I did not find any distortion being introduced by the input potentiometer, but maybe I was lucky or did not look hard enough. I concur with Mr. Bruinsma's recommendations on input potentiometers. If you can't get the ALPS potentiometer he recommends, at the least use a linear 50kΩ potentiometer of 24mm (or more) diameter (such as the Mouser 31VA405). This will make adjusting the "set level" more tedious for some signal levels.

DRIVER SPECS

After reading Mark Sanfilipo's article on Driver Parameter Measurement in the February 2005 issue of *audioXpress* magazine (p. 14), I was enthusiastic to try this technique, as I have an HP 54501 digital o-scope, a Wavetek function generator, and a number of drivers with unknown parameters.

I first set out to encapsulate the math into an Excel spreadsheet, but ran into a few problems. I'm not sure whether these are typos, inadvertent omissions, or just my lack of understanding.

1. In equation 7 you introduce a variable "m." It's not clear what this is since it is not listed in Table 1 or Figure 3, and is not calculated in any previous equation.
2. Equation 9 and Figure 3 reference variable "Yq"—the pre-pulse voltage. Isn't this just zero? I'm not sure why there would be any voltage across the speaker prior to the pulse. Also, it is not depicted in the figure (only T and I*Revc are

shown).

3. Figure 3 lists a variable "AN," which is described as the "voltage magnitude." However, it is not depicted on the figure and is not used in any of the equations. Is this just the peak-to-peak voltage (Y0-Y1)? The RMS voltage? Some other value?

Mathew Marulla
Nashua, N.H.

Mark Sanfilipo responds:

Great idea encapsulating the maths in an Excel spreadsheet!

1. "m" is an integer (0, 1, 2, 3, . . .) multiplier that, when plugged into equation 7, gives that position on the x-axis (in seconds) where maxima or minima occur.
2. Yq is the pre-pulse, quiescent voltage. Though ideally you should see this at 0, that isn't always the case. Its presence in the equation allows you to take into account any DC bias (if present) anywhere in the signal chain. Of course, if you can't be bothered to set the vertical position of the pulses as displayed on your oscilloscope so that they appear to launch from the x-axis, it can account for the display offset as well.

If Yq had been included in Fig. 3, it would have appeared between the X-axis and the horizontal bit of the trace immediately to the left of i*Re.

3. "AN" is actually "An," where n = 0, 1, 2, 3, . . . Take a peak voltage, subtract the pre-pulse voltage (if any), then take the RMS value of the result and you get "An."

SHOWCASE

I built the S-5 Selectronics K-12M tube amp (reviewed in Nov. '02 *aX*), included a home-brewed remote volume control, sunk the tubes, and added racing stripes in ebony and maple wood (*Photo 1*). Now in favor with the spouse, it looks good and has quality sound. The sparkle of the mid-range is very pleasing and rivals my surround system (at ten times the price).

My compliments to Larry Stafford of S-5 Electronics. Your publication is great. The best in electronics. Keep up the quality.

Rudy Dick
Abbotsford, B.C. Canada

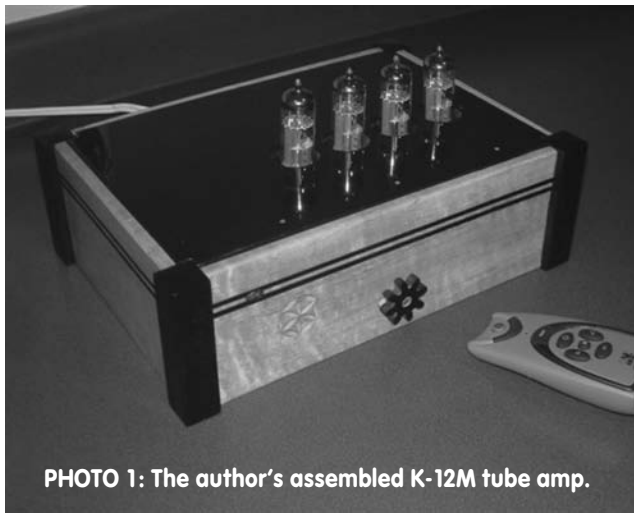

PHOTO 1: The author's assembled K-12M tube amp.

SIMPLE AMPS

As a long-time reader of *audioXpress* and previously *Glass Audio*, I would like to thank you very much for the excellent work you have done through the years. I think you have installed *Glass Audio* and also *audioXpress* as "The Authority in Tube Technique" all over the world.

I must say that the articles of Larry Lisle are especially very interesting and correspond very well with my interests in minimalist amplifier design. With his "Amplification Without Distortion" (July '03 *aX*), "A Minimalist Amplifier Design" (March '05 *aX*), and "A Battery Tube Hifi Amplifier" (*GA* 3/97) he described very simple amplifiers and gave very good and clear detailed descriptions of the techniques used. His explanations gave insights into the secrets of these simple amps which are no longer available.

Also, his explanation of the filament polarity (in "Amplification Without Distortion") and the (not intended?) negative feedback which results are well worth reading. These details prove that things are not really as they appear at first glimpse. Thank you very much, Mr. Lisle!

But, I have some questions, for which I was not able to find really

satisfying answers. Maybe Mr. Lisle can help. In some of his articles the windings of the transformers are additionally loaded with resistors. Also, in a German textbook with similar amplifiers of the 1920s, I have found capacitors across the primary windings of all transformers. The description of those amplifiers gave no explanation. It only said that it is sometimes possible to remove the capacitors without any effect.

What is the purpose of these resistors and capacitors? In which cases should a capacitor or a resistor be used?

Ulrich Tönjes
Bremen, Germany

Larry Lisle responds:

Thank you for your very kind words! I'm glad you have enjoyed my articles.

As to your questions, capacitors may be used with transformers for several reasons, one of which is to keep the amplifier from oscillating at either radio or audio frequencies. This could be a problem with the unshielded transformers that were sometimes used in the '20s. I suspect that this is the reason for their use in the textbook to which you refer, because the physical layout or shielded transformers might make their use unnecessary. Reversing the primary leads was another solution, as in "Amplification Without Distortion."

If the amplifier was to be used in a radio circuit following the detector, a capacitor might be used to bypass radio frequency energy to keep it out of the amplifier. Capacitors could also be used to shape the frequency response, either to peak it for voice or code reception in a communication receiver or to enhance the bass response.

Resistors are also sometimes used with transformers to change the frequency response. A resistor across the secondary reduces the value of Q of the resonant circuit formed by the leakage reactance and stray capacitance in the windings. This gives a more uniform high frequency response and also extends the low frequency response. A resistor across the primary extends the low frequency response.

Disadvantages of using "loading" resistors can include increased distortion, loss of gain, and resistor noise. The quality of the transformer, the circuitry, the impedances involved, and the purpose of the design influence whether or not loading resistors are needed, and their value. For instance, I used a resistor across the secondary of the excellent Hammond 808 transformer in "A Beginner's Push-Pull or Single-Ended Amplifier" (Jan. '02 *aX*), but not in "A Minimalist Amplifier Design."

For more information on resistor loading, see *The Radiotron Designer's Handbook,* 4th Edition, Chapter 5, Section 3, iii and Chapter 12, Section 4, xi.

SUITABLE OP AMPS

I follow Gary Galo's articles in *audioXpress*, and I want to thank him for sharing his knowledge. I have a crossover that uses the BB OPA2134 dual op amp, and I noted his preference for the AD745. Can you recommend a drop-in replacement for the OPA2134 that would be closer to the sound quality of the AD745? How about the AD827 or 828?

David Elderton
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Gary Galo responds:

Thank you for your nice comments. My preference for the AD745 is strictly in phono preamplifiers. The AD745 was designed for high-gain applications where low noise is essential. This op amp is only stable at gains of +5 or higher (or -4 for inverting topologies). It isn't unity-gain stable, and consequently is not suitable for electronic crossover or preamp line stage applications. Analog Devices does make a unity-gain stable op amp in the same family, the AD743. But, the AD743 has a very low slew rate of 2.8V/µS and a limited bandwidth of 4.5MHz, so I would not use it in high-end audio applications.

Since the AD745 is a FET-input device, with very low input bias current, it is suitable for situations where the source impedance is subject to change. Output offset must be maintained within certain limits with cartridges of varying impedances, or with no cartridge connected. This rules out such stellar performers as the AD797, which would otherwise be a first choice for phono preamplifiers. I know of at least one commercial phono preamp that uses the AD797, but its phono input is transformer-coupled, giving the AD797 a stable low-Z source regardless of the cartridge.

The OPA2134 is the dual version of the OPA134, and is already one of the best op amps made today for line-level applications, and certainly one of the very best dual

devices. Another chip you might try is the OPA2604, the dual version of the OPA604. Although Burr-Brown/TI boasts lower measured THD for the OPA134 family, I don't necessarily consider it sonically superior to the OPA604 family of op amps. Compare them yourself to see which you prefer in the crossover application.

For electronic crossovers and preamp line stages, you really want a FET-input device with a combination of very low DC offset *and* low input bias current. This rules out the bipolar AD827 and the AD828, which is also designed for voltage gains of +2 or higher.

The AD825, recommended in my articles "A Music Library Preamp" (March '05 *aX*) and "GFP-565 Preamp Follow-Up Mod" (Dec. '04 *aX*), is still the best FET-input op amp I've auditioned for line-level applications. Unfortunately, Analog Devices doesn't seem to have a dual equivalent of this chip, and it's only available in the SOIC package.

I don't generally use dual op amps, only because they may limit your upgrade choices down the road. There are a number of excellent dual op amps available, but I would normally rather keep all options open for upgrades when designing a project. That's my personal preference, of course.

HELP WANTED

I own a pair of L.M. Ericsson (Ericsson Telephones) Type 3 monoblock valve amplifiers for hi-fi use. They were used by the Australian Broadcasting Corporation in the 1960s.

They run two EL34s each and produce around 50W, I believe. What I'm wondering is whether anyone has ever heard of these amps and whether they know where I might find a circuit diagram for them.

I've tried Ericsson but, of course, they have no interest in helping.

If you've got any clues at all, I'd really appreciate the help.

Julian Mitchell
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New Zealand

I have been a *Glass Audio* subscriber in the past and am currently a subscriber of a*udioXpress*. I am very content with the magazine.

I have a Dynaco Series II Stereo 70 amplifier. I would like to know how to make my output tubes EL34/6CA7 last longer.

This series II Dynaco amplifier has a full wave, solid-state rectifier that applies full B+ voltage even before the tube filaments have had a chance to fully warm up. Because of full B+ voltage, cathode stripping has me concerned. I need to know exactly where the standby switch can be wired in on the secondary side of the power transformer. I have seen different variations of how the standby switches are wired in on other amps, but they seem to vary in the circuitry.

Can you show me a complete schematic diagram of how the standby switch can be wired in with "cushioning capacitors"?

Mario Montemurro
lpolsson@sd52.bc.ca

Two articles on this topic are in preparation. —Ed.

Readers with information on these topics are encouraged to respond directly to the letter writers at the addresses provided. —Eds. *aX*